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INTERNET

VoIP: a telephone conversation chopped into little packets

Greg decided to call Agnieszka because she had answered **his message** with only one sentence. At the Telephone Museum he found a rotary phone, stuck a finger into the hole marked nine and started turning the dial. “Hello? Why is this taking so long?!” “Greg, you're using a telephone from an age when people were expected to have patience.” ☎️🤣

A finger in the dial, no screen required

— Please call Agnieszka — Greg told the museum guide.

— Then dial the number.

— Dial? Where's the crank?

— Put your finger in the hole, turn to the stop and release.

Old rotary phones used **pulse dialing**. The dial mechanically interrupted the line circuit a certain number of times; the telephone exchange counted the pulses to identify each digit. Those were neither Internet packets nor DTMF tones. Later push-button phones could use **DTMF**, with a pair of audio frequencies representing each key.

— So the digit nine means nine interruptions?

— In a typical pulse system, yes. Zero normally means ten.

— No wonder phone numbers were shorter back then!

— That's not why, Greg.

It's important to distinguish **how a number is dialed** from how a conversation is carried. An analog telephone could use either a rotary dial or DTMF buttons. Replacing the dial with buttons didn't turn a call into VoIP.

When a wire really was a wire

The guide placed a copper cable on the table.

— Traditionally, your voice became a varying electrical signal on an analog subscriber line.

— And the whole conversation traveled along that one wire?

— Often on the local subscriber loop. The wider telephone network might already have been digital.

In conventional public telephony, exchanges established calls, with a circuit-switched channel assigned for the conversation. **An analog subscriber line didn't mean every exchange and long-distance trunk was analog.** Digital switching and transmission developed long before Internet telephony became widespread.

— Old cable, modern exchange?

— Precisely.

— What if somebody didn't answer?

— You heard ringing or busy tones, not an animation with three little dots.

Greg glanced at the dial.

— There was something honest about that. No “Seen” status.

VoIP: voice in little parcels

Next door, a young technician started a telephone connected to a computer network.

— Now for **VoIP, Voice over IP**. We turn speech into digital samples, encode it and transport it over an IP network.

— Cars on the packet highway again?

— This time, tiny parcels of sound.

VoIP doesn't require a traditional dedicated telephone circuit between exchanges. Voice data travels as packets, and the endpoints reconstruct a continuous audio stream. This can use an IP desk phone, laptop app, an analog telephone adapter or an operator's telephone service.

— Can I connect my old phone to the Internet?

— With an appropriate ATA gateway, if it supports what you need. Pulse dialing support isn't guaranteed.

— My rotary dial doesn't understand IP?

— It understands mechanics, which is fine.

SIP arranges the meeting, RTP carries the voice

At the VoIP exchange, Ms SIP sat behind a desk while a courier called RTP ran around carrying parcels.

— I want to call somebody! — Greg announced.

— I'll handle signaling — SIP replied. — We'll locate the other party and negotiate the session.

— And I transport the audio stream — added RTP.

SIP is a common signaling protocol for establishing, modifying and ending communication sessions. **RTP** frequently carries audio packets during a call. They aren't mandatory components of every VoIP product; other protocols and architectures exist.

— So SIP doesn't carry the actual conversation?

— Usually not. It organizes the connection.

— And RTP doesn't make the phone ring?

— Different job.

— A receptionist and a courier.

— With slightly fewer stamps than the post office.

Session setup often includes exchanging supported codecs and media addresses.

A codec packs human complaining

Greg spoke into the microphone: “Hello, hello, can you hear me?”

— Now we encode that — explained the technician.

— What if there's not enough room?

— We choose a **codec**. Different codecs have different bitrates, delays and compression methods.

G.711 is a straightforward traditional telephony codec; Opus adapts well to many Internet communication needs. Mobile networks also use codecs from the AMR and EVS families. What you hear depends on the entire call path, not just the number of megabits on your connection.

— I can have fast Internet and terrible sound?

— Absolutely.

— So I complain to my provider?

— Diagnose first. That's what you always do with servers.

— DON'T LECTURE ME WITH MY OWN METHODS!

Jitter, ping and missing syllables

The RTP couriers arrived one by one. Number three arrived before number two, and number five never turned up.

— He… llo… Gr… eg… — crackled the speaker.

— WHAT KIND OF ROBOT IS THAT?!

— **Latency, jitter and packet loss** — replied the technician.

High latency makes turn-taking awkward. Jitter is variation in packet arrival time, while loss can remove fragments of sound. A receiver can use a jitter buffer to smooth out irregular deliveries, but a larger buffer adds delay.

— What if I add more megabits?

— Bandwidth alone won't fix every timing or loss problem.

— But my speed test says five hundred megabits!

— A call needs predictable delivery of relatively little data.

— A broad highway with late couriers?

— That's why we measure path quality, not just road width.

NAT: the doorman doesn't know which room to call

Greg stood behind a router that let all the residents out through one gate.

— Send the audio to my phone! — he demanded.

— Which one? — asked the NAT doorman.

— MINE!

NAT and firewalls can make VoIP awkward. A device may know its private address, but somebody outside the network might not be able to send audio packets directly to it. Call signaling can work even when sound travels only one way, or not at all.

— I can dial somebody but hear nothing?

— Yes. That's a classic symptom of some media-path problems.

STUN helps a device learn how its address appears from outside, **TURN** can relay media, and **ICE** helps select a workable path from available candidates. These mechanisms are particularly familiar from WebRTC. Traditional SIP systems also use other techniques, such as suitable server, gateway and session-border configuration.

— Why three helpers?

— Different problems.

— Can I just turn NAT off?

— Not always, and not everywhere. Don't redesign the whole network because you can't hear “hello”.

The mobile phone that remembers the old days

A technician walked into the museum carrying a brick-sized mobile phone.

— I remember life before LTE!

— So all mobile calling was analog back then? — asked Greg.

— Mind the generations.

Early mobile systems such as **1G used analog voice**. Later **2G systems such as GSM were already digital**, as was 3G, although conventional mobile voice in those generations generally used the circuit-switched domain rather than ordinary application IP calls.

— GSM wasn't analog?

— Correct. An important difference.

— Why did my phone sometimes show 4G, then switch to 3G or 2G during a call?

— That may have been **CS Fallback**: voice moved to an older circuit-switched service when VoLTE wasn't being used.

— And now?

— Now for something that sounds like VoIP, with its own framework.

VoLTE isn't WhatsApp wearing a different hat

The technician held up a modern smartphone.

— **VoLTE, Voice over LTE**, is an operator-provided packet voice service over LTE, generally based on the **IMS** architecture and SIP signaling.

— So it's VoIP?

— In the broad sense, it is voice over IP. But it's a **mobile operator's telephone service**, with standardized call handling, telephone numbers and quality mechanisms, not merely a random Internet calling app.

VoLTE can provide faster call setup and good voice quality, depending on devices and networks. The phone need not drop from LTE to an older radio technology just to carry a call, as happens with CS Fallback. 5G also has a path for voice service, including VoNR, but not every network and phone uses it in every situation.

— So 1G was analog, 2G and 3G digital but usually circuit-switched for voice, and LTE with VoLTE carries voice in IP packets?

— Exactly.

— Then why did we need the whole lecture?

— So you wouldn't describe GSM as an analog rotary phone with an antenna.

— That would have sounded wonderful, though.

Making a call takes more than talking

Greg picked up the handset again.

— Does any of this work without electricity at home?

— It depends on the equipment and infrastructure. Traditional analog telephones often drew power from the exchange over the line, while IP phones, routers and switches generally need local power or suitable backups.

— So an old phone might work through a power outage?

— If the exchange and line were operating and the handset needed no local adapter, often yes.

— And VoIP?

— Consider a UPS and reliable connectivity, especially for important calls.

— New doesn't always mean simpler, then.

— It means different abilities and different dependencies.

Actual call quality and reliability depend on devices, network, configuration, power and the service provider. No codec can repair a disconnected cable.

Hello? Finally, I can hear you!

Agnieszka answered the call.

— Hello?

— AGNIESZKA! FINALLY! — Greg shouted.

— What happened?

— I wanted to check whether you got my message.

— I did. I replied.

— But I wanted to hear you say it.

— Now you have.

VoIP turns speech into data carried across an IP network. Signaling helps set up a call, codecs prepare sound and packets need to arrive regularly enough for intelligible speech. Old analog telephony, digital circuit-switched networks, modern VoIP and VoLTE solve much the same human problem in different ways.

— Can I call again to confirm I heard you?

— Greg, no.

— A text message, then?

There was silence at the other end. The engineers checked the jitter. This time it was perfect.

The Telephone Museum and Packet Exchange management reports: the rotary telephone works, the SIP switchboard accepts calls, the RTP couriers are arriving in order, and the NAT doorman has received new instructions. Greg dialed the number nine times because one turn of the wheel “didn't seem like enough”. Agnieszka remains technically reachable. Organizationally, not necessarily. ☎️📞🤣❤️