TDM vs VoIP

This article will detail the difference between TDM (Time Division Multiplexing) and VoIP (Voice Over Internet Protocol).

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Before describing the issue, here is a definition of terms that are used within the document:

**PSTN** – *(Public Switching Telephone Network)* – Is the world's collection of interconnected voice-oriented public telephone networks, both commercial and government-owned.

**POTS** – *(Plain Old Telephone Service)* - Voice-grade telephone service employing analog signal transmission over copper loops. Residential phone service and analog based PBX’s used this type of service.

**TDM** – *(Time Division Multiplexing)* - is a method of transmitting and receiving independent signals over an analog signal path.

**SS7** – *(Signaling System No. 7)* – It is a set of telephony signaling protocols developed in 1975, which is used to set up and tear down most of the world's public switched telephone network (PSTN) analog telephone calls.

**VoIP** – *(Voice Over Internet Protocol)* – It is a methodology and group of technologies for the delivery of voice communications and multimedia sessions over Internet Protocol (IP) networks, such as the Internet. The terms Internet telephony, broadband telephony, and broadband phone service specifically refer to the provisioning of communications services (voice, fax, SMS, voice-messaging) over the public Internet, rather than via the public switched telephone network (PSTN).

**SIP** – *(Session Initiation Protocol)* – It is a communications protocol for signaling and controlling multimedia communication sessions. The most common applications of SIP are in Internet telephony for voice and video calls, as well as instant messaging, over Internet Protocol (IP) networks.

**CID** – *(Caller ID)* - is a telephone service, available in analog and digital phone systems and most voice over Internet Protocol (VoIP) applications, that transmits a caller's phone number to the called party's telephone equipment during the ringing signal, or when the call is being set up but before the call is answered. The caller's phone number is passed along in the signaling message.

**CNAM** – *(Caller ID Name)* – A service provided within the PSTN to deliver a name with the Caller ID. Where available, the terminating carrier of a phone call can do a lookup on the Caller ID through a 3rd party service or with the carrier of...
record to determine a name associated with the calling telephone number.

For years, the PSTN was synonymous with POTS as all carriers were using analog voice networks (TDM) for both internal connections and external connections to other carriers. This interconnection of carriers is what makes up the global PSTN network. This network has been around in basic form since the inception of telephone service. In the early 2000's, VoIP was introduced within Voice Networks but just on the end points. VoIP phones were introduced but the VoIP connection only extended to the PBX. From the PBX it was still traditional TDM circuits (T1, PRI, etc.) back into the PSTN. Over the years since, carriers have been replacing older TDM circuits within their own networks with SIP/VoIP circuits. This allowed for more calls over the same pipe (through digital compression) and saved costs. To do this, carriers had to utilize gateways throughout their networks to convert between TDM and VoIP networks. (Analog to Digital conversions). While the gateway gave the carriers the ability to originate or terminate calls on either network, if your call passed through too many gateways, call quality would suffer after so many conversions.

Connections between carriers were mostly still TDM circuits with traditional PSTN services. Over the past couple of years however, more and more carriers have been changing their interconnects between each other to be SIP/VoIP as well, thus eliminating the need for gateways. While this improves efficiency by eliminating gateways, it does change how some traditional PSTN services (Network Announcements, CNAM, etc.) are delivered. Also, many of these services are not regulated or standardized so carriers have the ability choose how they want to (if at all) deliver those services. As an example; in the past before smartphones, wireless providers did supply CNAM (The name associated with the Caller ID) to cell phones. However, with the advent of smartphones and the phone matching the caller id with a contact, wireless providers in general no longer supply CNAM information to phones. This saved them money on CNAM dips. Instead they just now provide City and State.

CNAM was a feature of the traditional PSTN network that was delivered over the SS7 portion of the network. The terminating carrier would receive the call, get the caller id and then query over the SS7 back to the originating carrier or national database to get the associated name and deliver that to the end caller. All carriers were on SS7 for communications and this worked pretty well. All of them updated the same national databases and everyone queried the same way.

With VoIP, networks changed. You now had two different networks running side by side with different capabilities. While SS7 is the signaling protocol for TDM networks (analog), SIP is the signaling protocol for VoIP networks (digital). SIP has more information contained within its message than SS7 does and this information would get stripped out when the call passed through a gateway onto a TDM network. With carriers now turning up more and more SIP interconnections, the possibility of a call being SIP from end to end is increasing. At some point in the future, carriers will have removed all TDM based circuits (their goal) and we'll be back to a single Voice Network running on SIP/VoIP.

With SIP comes additional capabilities. As part of the information packet that is sent when a call is placed, the name of the user who is calling is passed. If the call is SIP from end to end, then the terminating carrier will receive this information and will display this information for CNAM. If the user name is blank (which would happen if it passed through some gateway along the way. Again, this information would have been stripped out in the gateway conversion), the terminating carrier would then choose to either display just city and state or dip the national CNAM database. The carrier can also dip the national CNAM database and cache that information for a period of time so that subsequent calls
from the same number doesn’t require them to pay the dip charge again. This can cause delays in CNAM updates being seen on end user’s phones. Terminating carriers can also choose to use their own database and gather the data on their own and not dip the CNAM database. AT&T has announced their intention to do this and have already started on their own Universal Caller ID.

There are currently two providers of a national CNAM database. VeriSign and Targus (Neustar). For the databases to be accurate, it requires that all DID providers act in good faith and send CNAM updates to these companies. A DID provider can choose to update one, both or none of them. The database providers also gather information from other sources but the main source of information is from the DID providers. Industry analysis has shown these databases to be badly out of sync with carrier information.

Because of these complexities and costs for SIP, some terminating SIP carriers are choosing the path of the wireless providers and not provide CNAM. Also with the advent of more VoIP platforms like Microsoft’s Skype for Business, the need for CNAM is going away. With a Skype for Business call where it is from one Skype user to another, this communication is direct between users with no PSTN involved and therefore no CNAM. Any information displayed would be based on the users contacts and user name.