

## **Ever thought about extending circuits over IP?**

As circuit-switched operators migrate to IP networks, one way of preserving the range and quality of legacy PSTN services is to opt for TDM over IP.

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High-speed IP-based networks are the latest innovation in the world of communications. The capacity of these networks is increasing at a prodigious rate, fuelled by the popularity of the internet and decreasing costs associated with the technology. Worldwide data traffic volume has already surpassed that of the telephone network, and for many applications, the pricing of IP traffic has dropped below the tariffs associated with traditional TDM service. For this reason, significant effort is being expended on VoIP technologies.

Inherent in all forms of VoIP is revolutionary change, whereby much of the existing telephony infrastructure will be replaced by novel IP-based mechanisms. Despite the hype, this effort has been more protracted and less successful than initially expected. Today's telephony technology (both those portions that VoIP aims to replace and those to which VoIP must interface) is extremely complex. Revolutionary implementations of its hundreds of features and thousands of variations cannot be expected to be developed in a short time frame.

There is, however, an alternative method of exploiting IP networks for telephony service which is evolutionary rather than revolutionary. This method uses IP networks as a drop-in replacement for native TDM networks. It seamlessly interfaces to all existing equipment, such as legacy PBXs and switches, and inherently provides all the hundreds of telephony features and the PSTN quality to which customers have become accustomed. This alternative is circuit extension over IP using TDM over IP.

### **The troubles with VoIP**

In principle it's not that difficult to carry voice over IP networks; a digitised voice signal is simply data and can be carried by a packet network just like any other data. The major technological achievement of the telephone network, that of least-cost routing, has its counterpart in IP networks as well.

There are, however, two fundamental problems that have to be solved before VoIP can realistically compete with TDM networks, namely QoS and signaling.

The meaning of Quality of Service is completely different for data and voice. Although most data can withstand relatively significant delay, low delay and proper time ordering of the signal are critical for voice applications, even though the loss of a few milliseconds of signal is usually not noticeable. These requirements are completely at odds with the basic principles of IP networks (although not necessarily with those of other packet networks).

To overcome these constraints, mechanisms such as tunneling and jitter buffers need to be employed. Additional components of voice quality, such as echo cancellation and voice compression, are not inherent in data-based networks at all and need to be added *ad hoc* for VoIP.

And because almost all of the massive R&D effort in the field of VoIP is directed towards solving the QoS problem, the signaling problem has remained largely unsolved. By signaling, we mean the exchange of information needed for a telephone call other than the speech itself. Signaling allows basic features such as informing the network that the phone is off-hook or it needs to ring; more advanced features, courtesy of signaling, make sure the call reaches its proper destination and the call is billed accurately. More sophisticated applications enabled by signaling are caller identification, call forwarding, and conference calls. There are literally thousands of such telephony features, with dozens of national and local variations. Phone customers are mostly unaware of this complexity, at least until you try to deprive them of any of the features to which they have become accustomed.

The problems arise when one must interface between the IP network and the standard telephony network, a connection that is imperative in view of the universal availability of standard telephone sets.

## **TDM over IP**

**TDMoIP** transparently transports the TDM frame without any attempt at interpreting the data. It is completely oblivious to such TDM internals as time slots and signaling channels. Thus **TDMoIP** can be used to transport arbitrary T1/E1 services, even if some of the channels are actually used to carry data or if the entire frame is an unstructured bit-stream.

Similarly, the basic **TDMoIP** concept is easily extended to fractional T1 or channelised E1 systems. To reduce traffic, only the information carrying bytes needs be included in the IP packet.

But how does **TDMoIP** solve the signaling problem inherent in interfacing between IP networks and the telephony network? **TDMoIP**'s in-band signaling, as its name implies, is transferred in the same audio band as the speech. It can take the form of aural call progress tones that are encoded in the TDM timeslot and automatically forwarded by **TDMoIP**. Speech compression algorithms used by VoIP systems often do not pass these tones well. Therefore, VoIP systems need to implement tone relay protocols to ensure that in-band signaling functions properly.

## **Synchronisation**

Another function available in customary TDM networks is time synchronisation. The hierarchy of time synchronisation is essential for the proper functioning of the network as a whole.

Packets in IP networks reach their destination with delay that has a random component, known as jitter. When emulating TDM on an IP network, it is possible to overcome this randomness by using a buffer to smooth out all incoming data, assuming the proper time reference is available. For the most part, however, the original time reference information is no longer available.

In principle there are two different levels of integration of **TDMoIP** into the telephone network. In the toll bypass scenario, a competitive carrier introduces an alternative link, based on **TDMoIP**, between two central offices. Due to cost advantages, 'toll bypass' services can be offered to customers at a lower rate than the incumbent's rate. In such applications, both **TDMoIP** devices should receive a time reference from the central offices to which they connect.

In the whole network scenario, major portions of the primary infrastructure are replaced with **TDMoIP** networks, and a method of time synchronisation is required. IP networks also disseminate clock information through a protocol known as NTP; but unless the IP network is completely private and dedicated to the **TDMoIP** link, there will be no connection between the NTP clock and the desired TDM one. One solution is to provide time standards, such as atomic clocks or GPS receivers to all **TDMoIP** devices, thus relieving the IP network of the need to send synchronisation information. When the provision of accurate local time references is not possible or too costly, then recovery is possible when the destination regenerates and synchronises the clock.

## Pros and cons

**TDMoIP** is simpler than VoIP because it is transparent to voice and data signaling and protocols, even when they are proprietary. VoIP, on the other hand, still has issues with new protocols and translation between signaling formats. VoIP does hold the promise of new applications, whereas **TDMoIP** automatically takes advantage of existing PBX and CTI features. As far as bandwidth optimisation, VoIP gateways include DSPs which allow voice compression and silence suppression and therefore reduce bandwidth requirements. However, this is at the expense of lower quality and higher latency.

The simplicity of **TDMoIP** translates into lower cost of ownership with initial expenditure savings and operational cost benefits. These savings are significant for the enterprise consumer who is typically not interested in replacing his legacy TDM equipment and dealing with 'forklift upgrades' with the associated training and maintenance expenses. Another important difference is that due to transparency, **TDMoIP** can offer mixed voice-data services.

From a service provider point of view, **TDMoIP** and VoIP complement each other. Extending TDM trunks over the IP network transparently from the customer site to the carrier POP makes it simple for the carrier to deploy larger, scalable VoIP gateways and soft switches at the POP where resources are available, and provide the user with a simple **TDMoIP** Network Termination Unit (NTU) at the customer premises. These **TDMoIP** circuits could then be used to provide a number of services in addition to VoIP, such as regular PSTN access, Centrex, frame relay and ISDN.

## ATM and gigabit ethernet

**TDMoIP** provides many of the benefits of ATM in that end-to-end delay starts at less than 2 ms and integrity of structured or unstructured T1s or E1s is maintained. **TDMoIP** is much simpler, cheaper and more efficient than ATM. Most importantly, it can be carried over IP and ethernet networks. **TDMoIP** is more efficient than ATM because its payload size and therefore percentage of overhead can be set on a per-application basis. This is possible because the number of octets per frame is configurable. With ATM, payload is always 48 bytes, and therefore the percentage of overhead is much higher.

Gigabit ethernet (and 10-gigabit ethernet) is being used in metropolitan area networks (MANs) and wide area networks (WANs). In particular, gigabit ethernet over dark fibre is becoming a popular alternative to SONET and ATM. However, ethernet is basically a data network technology and cannot by itself handle voice traffic. **TDMoIP** empowers gigabit ethernet with voice and circuit extension capabilities and therefore is an excellent complementary technology. Together these technologies are expected to increase market share by offering a simpler, less expensive alternative to VoIP, ATM and SONET. History shows that the simpler, cheaper technologies such as ethernet and frame relay tend to prevail over the more complex and expensive alternatives such as FDDI, ATM and token ring, even if the latter are more robust.

Initial concerns regarding simpler technologies like gigabit ethernet are usually rapidly put to rest by subsequent enhancements. For example, a SONET ring topology is considered very reliable because of its rapid recovery from a failure or fibre cut. Gigabit ethernet does not inherently have this capability, but an alternate trunk can usually be switched within a few milliseconds. Even if there is only a single fibre between the switches, protocols such as OSPF enable routing tables to be updated within a few seconds, and the IP data stream to be quickly reconnected.

Another important example relates to QoS, where ATM has the most defined service level categories. However, today's gigabit ethernet switches and terabit routers implement advanced mechanisms to prioritize packets and reserve bandwidth for specific applications. By marking **TDMoIP** packets (using 802.1p&q, ToS, and set UDP port numbers) they may be easily identified and prioritised.

Combining **TDMoIP** with gigabit ethernet switches and terabit routers ensures a worthy alternative to SONET and ATM, providing a simpler, more cost-effective solution with more bandwidth and more efficient grooming of T1s or E1s to gigabit optical networks.

## **The carrier market**

Competitive local exchange carriers (CLECs) can take advantage of **TDMoIP** technology to expand their market share quickly, simply and inexpensively with more points of presence (POPs) and more services. Incumbent local exchange carriers (ILECs) providing basic transport of circuits wholesale to the CLEC can use **TDMoIP** to extend the T1 or E1 circuits cost-effectively over optical or wireless links back to the CLEC service POP.

Some CLECs or utility companies will also be able to increase their revenues by taking advantage of their wireless or optical data networks and use **TDMoIP** to extend T1 or E1 circuits between an entity and facilities at another location. In this way, the service provider can bundle private line services in addition to broadband data. Bundled services could include:


- high-speed internet access, web hosting and VLAN;
- traditional PSTN access, or centrex service, by extending circuits from the class 4/5 switch at the CO to the customer site;
- new voice and unified messaging services using VoIP gateways or switches and extending circuits to the remote site;
- private line services; and
- local loop access to ATM, frame relay, ISDN and X.25 networks.

**TDMoIP** offers the CLEC an alternative method of extending services to its customers. Traditionally it was necessary to lease the local loop from the incumbent local exchange carrier (ILEC), but now any packet network will do. All voice and data services that traditionally run on T1 or E1 circuits are automatically supported over IP. This not only includes simple PSTN access, but also PRI, Centrex, and VoIP for voice services and ATM, frame relay, PPP, ISDN, SNA, X.25 for data services. The packet networks are also no longer limited to synchronous ATM and SONET, but can run equally well on 10/100BaseT wireless radios, gigabit ethernet, or any other network.

## Conclusion

Circuit extension over IP using **TDMoIP** technology provides the benefits of IP networks without the dangers of a new technology. It supports legacy telephony switches or PBXs over data infrastructures with no loss of functionality. In addition, it empowers IP networks with leased line services for voice, ATM, frame relay, SNA, and so on. Finally, the fact that it's simple, inexpensive and transparent to all signaling and protocols makes it an ideal complement to gigabit ethernet in next generation networks. ➡

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