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NEWTON's TELECOM DICTIONARY

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NEWTON'S TELECOM DICTIONARY

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Help Make This Dictionary Even Better

by Harry Newton

I wrote this book for those of us turned on, for those of us trying to keep up, for those of us new to this wonderful industry and for those of us who simply want a respite from life for a few moments.

Give my dictionary to your employees, your users, your customers, your boss. Give it to your kids to let them understand what you do and why you're turned on. They'll understand why you, too, have no life.

Dip into this book. Dig around. You'll find something interesting, something you didn't know, something irrelevant, something relevant, something educational, something amusing, something warming. I have fun writing this dictionary. You should have fun reading it.

This is a dictionary to work with every day. Companies bring their new employees up the telecom and networking learning curve. Salespeople include the definitions in proposals to customers. Users explain telecom things to their boss. Lawyers even use it in court. Sometimes they rely on it. God help us.

Most technical dictionaries define terms tersely, often in other technical terms. They leave you more confused. This dictionary is different. I define the term, talk about how it works, how you use it, what its benefits are, what its negatives are, how you should go about buying and installing it and some warnings.

I'm adding, updating, expanding and fixing 100 words a month. Yet, neither Ray Horak, the best technical contributing editor ever, nor I can keep up. We beg you. If we're missing a definition, or if a definition is unclear or wrong, please email me. Harry@HarryNewton.com.

I'm looking for corporate glossaries and dictionaries. I don't want ones that just have your company's products, nor ones that are secret. I want ones with generic definitions of the specialized technical areas your company is in. Please don't send acronyms. There are more stupid acronyms in this world than Carter had liver pills. If I included all the acronyms I get sent, this poor dictionary would be 5,000 pages. Moreover, each acronym would have ten totally different meanings. So you'd never know.

If you send me a glossary or several new, legitimate (not made-up) definitions I can use, I'll reward you with a free copy of the next edition. Please send me your street address.

Thank you.

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P.S. Arthur Clarke once commented "Any sufficiently advanced technology is indistinguishable from magic."
"I hope the year 2002 brings lots of magic into your life."

Flexibility is achieved by means whereby the essential requirements applicable to new network termination types can be determined in a timely manner. The essential requirements are restricted to:

- (a) Prevention of misuse of public network resources causing a degradation of service to third parties.
- (b) Interworking via the public network(s) and Community-wide portability between ONTPs specifying a basic level of interworking, e.g. simple voice telephony but excluding supplementary services.
- (c) Effective use of spectrum allocated to terrestrial/space radio communication and used for radio services recognizing that trade-off will be necessary between the quality, capacity, and availability.

For each type of Connected Terminal Equipment (CTE) formerly defined as Telecommunications Terminal Equipment, the essential requirements applicable are to be selected from a master list contained in the Directive. The technical requirements will be defined in appropriate technical specifications. These will be harmonized European standards or, in cases where such standards do not yet exist, other appropriate technical specifications. The specifications of essential requirements will take into account the following additional requirements for the common good:

- (a) Protection of health, e.g. minimizing the health hazards of radio frequency radiation.
- (b) Features for users with disabilities.
- (c) Features for emergency and security services.
- (d) Protection of individual privacy.

CTI Computer Telephone Integration. A term for connecting a computer (single workstation or file server on a local area network) to a telephone switch (a PBX or an ACD) and having the computer issue the telephone switch commands to move calls around. The classic application for CTI is in call centers. Picture this: A call comes in. That call carries some form of caller ID — either ANI or Caller ID. The switch “hears” the calling number, strips it off, sends it to the computer. The computer then does a lookup for the numbers in a database, sends the switch back instructions on what to do with the call. The switch follows orders. It might send the call to a specialized agent or maybe just to the agent the caller dealt with last time. Meantime, the agent sees a screen pop of information about the caller — such information having been pulled up out of the server’s database, using the caller ID information.

CTI and CT (computer telephony) are often confused. In fact, CTI is the older and smaller term. CTI has been the dismal part of computer telephony

— the difficult integration of reluctant, closed phone systems with outside computers to which they were never meant to talk. Computer telephony (or CT) is more exciting because it’s building new phone systems with fantastic new features based on open standards, open hardware and open software. CTI covers integration with switches. CT covers that AND a lot more — like callback, the UnPBX (communications server), the central office in a PC, IP telephony, one number find me, predictive dialing, unified messaging, interactive voice response, fax blasting and serving, etc.

See also Computer Telephony, TAPI, TAPI 3.0, TSAPI and Windows Telephony.

CTIA Cellular Telecommunications & Internet Association, aka Cellular Telecommunications Industry Association. The name change was made to reflect the increasing focus on wireless Internet access. Based in Washington, D.C., the CTIA is a trade association representing the interests of the wireless telecommunications industry. CTIA also is the parent of CIBERNET Corp., which provides the cellular industry with inter-company billing protocols, roaming administration tools, and financial settlement programs. www.ctia.org.

CTIP Computer Telephony Interface Products. Adapters that allow telephones to work with computers. An example is the Konexx connector, which fits between the handset and the phone, and allows a connection to a PC modem or fax machine. This definition contributed by Larry Kettler of San Diego.

CTN 1. See Call Tracking Number.

2. Consumers’ Telecommunications Network. A telecommunications association in Australia that represents consumers and looks after their interests: the CTN is part of the Australian Communications Authority’s Consumer Consultative Forum (CCF). See Consumer Consultative Forum.

CTO See Chief Technology Officer.

Ctrl Control. The label on the control key on your computer.

CTS 1. Clear To Send. Pin 5 on the 25-conductor RS-232-C interface or an RS-232-C signal used in the exchange of data between the computer and a serial device. In short, Clear

to send is one of the nine wires in a serial port used in modem communications, CTS carries a signal from the modem to the computer saying, “I’m ready to start when you are.”

2. Communication Transport System. CTS is The Siemon Company’s proprietary structured wiring system. It consists of the methodology and the connecting hardware products to plan, design, and implement the communications wiring infrastructure for commercial buildings (for more information see the company’s CTS Design Workbook and CTS Training Videotape). The Siemon Company is based in Watertown, CT.

3. Conformance Testing Services.

CTTS Coax To The Curb. An approach that provisions a multiline remote terminal to deliver voice and data to concentrated residential applications.

CTTU Centralized Trunk Test Unit. An operational support system providing centralized trunk maintenance through a data link on a switch.

CTX Centrex.

CTY Console Teletype, a contraction of “Console” and “TTY” (Teletype). A terminal keyboard associated with the console of a computer system, such as a PBX. Also, the designation for the cable that connects the CTY to the computer system. See also Console and TTY.

CU-SeeMe A popular videoconferencing and videophone product that works over the Internet. CU-SeeMe was originally developed at Cornell University and is available free for the PC and the Macintosh. An enhanced commercial version that adds an electronic chalkboard is available from White Pine Software. The software is designed for personal use and for use in instruction and in business communications.

CUA Common User Access. The policy of using the same command for a given function in all software. This makes the software easier to learn and use because you only have to learn one set of commands. Windows has a set of CUA guidelines which many Windows programs follow. For example, Alt+F4 always means close this window.

Cube Farm An office filled with cubicles. See also Prairie Dogging.

Cuckoo-Clock Telecom slang for some 6- and 10-button models of AT&T 1A2 wall phones shaped vaguely like traditional cuckoo-clocks. These were probably the first multiline phones to come with handsets that plugged into the base with Trimline-style 5-pin plugs, before the current modular connectors were adopted. Often seen in hospitals on TV shows.

CUG Closed User Group. Selected collection of terminal users that do not accept calls from sources not in their group and that are also often restricted from sending messages outside the group.

Curie Point The temperature at which certain elements (usually so-called “rare earth” elements) relax their resistance to magnetic changes. In a magneto-optic disk drive the surface to be marked is heated briefly by a laser light to its Curie point. Magnetism is then applied in the proper polarity to make the spot a “1” or a “0.” It cools, and is locked in that position, until it is re-heated and changed again. This is how magneto-optic drives can be erasable.

Current A measure of how much electricity passes a point on a wire in a given time frame. Current is measured in amperes, or amps. The abbreviation for current is I. See Ohm’s Law.

Current Carrying Capacity The maximum current an insulated conductor can safely carry without exceeding its insulation and jacket temperature limitations.

Current Limit The function of a circuit or system that maintains a current within its prescribed limits. A circuit breaker terminates current flow when current exceeds the trip limit. Most UPS systems have an electrical subcycle current limit that regulates the output current to a value within the UPS design limits. This subcycle current limit may activate when a load demanding high inrush current (like a computer or phone system) is switched on. The activation of the subcycle current limit protects the UPS from damage but allows the output voltage to become distorted or even collapse momentarily. Most on-line UPS systems will have the subcycle current limit activated by computer load switching and use a bypass in order to maintain load continuity when the current limit activates. Standby and line-interactive UPS systems can draw on the utility grid directly to supply load switching current transients and therefore do not activate the subcycle current limit or need to use the automatic bypass feature. This definition from APC.

Current Loop Transmission technique that recognizes current flows, rather than voltage levels. It has traditionally been used in teletypewriter networks incorporating batteries as the transmission power source. In this serial transmission system, a pair of wires connecting the receiving and sending devices transmits binary 0 (zero) when no current flows and binary 1 (one) when current is flowing.

Cursor A symbol on a screen indicating where the next character may be typed. Cursors may be solid, blinking, underlines, etc. Many programs, computers and phone systems

the precise hookswitch signal for the precise length of time necessary — no more, no less. The Tap Button is also called a Flash button or a Tap Key.

Tap Key Also called Tap Button or Flash Key. A button on a phone that accomplishes the same function as a switch hook but is not a switch hook. See Tap Button.

Tap Loss In a fiber optic coupler, the ratio of power at the tap port to the power at the input port.

Tap Port In a coupler where the splitting ratio between output ports is not equal, the output port containing the lesser power.

TAPAC Terminal Attachment Program Advisory Committee. Body which recommends telecom standards to the Canadian Federal Government.

Tape Bomb Troubling news that scrolls across the bottom of CNBC and CNNfn, catching investors off guard and without time to sell (or buy).

Tape Drive The physical unit that holds, reads and writes magnetic tape.

Tape Reader A device which reads information recorded on punched paper tape or magnetic tape.

Tape Relay A method of retransmitting TTY traffic from one channel to another, in which messages arriving on an incoming channel are recorded in the form of perforated tape, this tape then being either fed directly and automatically into an outgoing channel, or manually transferred to an automatic transmitter for transmission on an outgoing channel.

Tapered Fiber An optical fiber in which the cross section, i.e., cross-sectional diameter or area, varies, i.e., increases or decreases, monotonically with length.

TAPI Telephone Application Programming Interface. Also called Microsoft/Intel Telephony API. A term that refers to the Windows Telephony API. TAPI is a changing (i.e. improving) set of functions supported by Windows that allow Windows applications (Windows 3.xx, 95 and NT) to program telephone-line-based devices such as single and multi-line phones (both digital and analog), modems and fax machines in a device-independent manner. TAPI essentially does to telephony devices what Windows printer system did to printers — make them easy to install and allow many application programs to work with many telephony devices, irrespective of who made the devices. TAPI is one of numerous high-level device interfaces that Windows offers as part of the Windows Open Services Architecture (WOSA). TAPI simplifies the process of writing a telephony application that works with a wide variety of modems and other devices supported by TAPI drivers. See also Dial String, Microsoft Fax, TAPI 2.0, TAPI 3.0, Windows 95 and Windows Telephony for fuller explanations.

TAPI 2.0 The following is Microsoft's explanation: The Microsoft Windows Telephony API (TAPI) 2.0 ships as part of Windows NT Server 4.0 and Windows NT Workstation 4.0. TAPI 2.0 is the latest release of the TAPI specification, introduced in 1993. TAPI helps bridge the gap between the telephone and computer. TAPI helps the PC to understand how telephone networks operate. With TAPI, programmers can exploit telephone network capabilities from within regular Windows-based applications. With TAPI, the jungle of proprietary telephony hardware is turned into standard programming interfaces. Application developers can wave goodbye to the complexity and variability of the underlying telephone network. They no longer have to hard-code their applications to a particular system's signaling or message set requirements. Instead, applications developers write code to the Telephony Applications Programming Interface. Developers can focus on their application without worrying about the nitty-gritty programming details of connecting to a specific telephone network. TAPI supports PBXs, key telephone systems, ISDN, the analog PSTN, cellular, CEN-TREX and other types of telephone networks. (So long as the various providers have written an SPI — Service Provider Interface, which is code to translate their commands into code which TAPI can understand. — Harry)

TAPI 2.0 includes these enhancements:

- 32-bit architecture. All core TAPI components are now based on the Win32 architecture. Non-Intel processors running Win NT Server 4.0 or Workstation 4.0 are supported.
- 32-bit application portability. Existing Win32 apps currently running on Win 95 using TAPI 1.4 will run on NT Workstation 4.0 or Server 4.0 on Intel x86 microprocessors.
- 16-bit application portability. Existing applications currently running on Win 95 and 3.1 using TAPI 1.3 will run on NT Workstation 4.0 or NT Server 4.0 on Intel x86 microprocessors.
- Unicode support. Win32 apps can now call the existing ANSI TAPI functions or the new Unicode versions of functions. Unicode is a 16-bit, fixed-width character encoding standard. It encompasses virtually all of the characters commonly used on computers today.
- Expanded feature support for call center applications. TAPI now supports an expanded set of features to better serve call center operations with Windows. New call center features

supported include ACD queues, predictive dialing, and call routing.

- Registry support. All telephony parameters are now stored in the Windows registry. All stored parameters can be updated across the LAN.

- Quality of Service (QoS) support. Applications can request, negotiate, and re-negotiate QoS performance parameters with the network. Improved QoS support reduces or eliminates latency and other negative characteristics for applications, especially voice and data apps, over various networks.

- Enhanced device sharing. Applications can restrict handling of inbound calls on a device to a single address. This supports features such as distinctive ringing.

- Additions and changes to TAPI functions. Many new TAPI functions and messages are available with TAPI 2.0. In addition, several functions and messages already supported by TAPI 1.4 were changed in some measure to make them more consistent in their operation. See TAPI 3.0

TAPI 3.0 Microsoft announced TAPI 3.0 in mid-September, 1997. At that time, Mitch Goldberg of Microsoft, said TAPI 3.0 will be available initially in NT 5 beta 1 in September, fully featured in NT 5 beta 2, and available commercially when the NT 5.0 operating system ships (around the middle of 1998). TAPI 3.0 will also be available for Windows 98 ("Memphis") in the NT 5.0 time frame. microsoft.com/communications

At the same time, Microsoft released a White Paper titled "IP Telephony with TAPI 3.0." Here are key excerpts. For the full paper, www.ctexpo.com. TAPI 3.0 is an evolutionary API providing convergence of both traditional PSTN telephony and IP Telephony. IP Telephony is an emerging set of technologies which enables voice, data, and video collaboration over existing LANs, WANs and the Internet. TAPI 3.0 enables IP Telephony on the Microsoft Windows operating system platform by providing simple and generic methods for making connections between two or more machines, and accessing any media streams involved in the connection. TAPI 3.0 supports standards based H.323 conferencing and IP multicast conferencing. It utilizes the Windows NT 5.0 operating system's Active Directory service to simplify deployment within an organization, and includes quality of service (QoS) support to improve conference quality and network manageability.

What is IP Telephony? IP Telephony is an emerging set of technologies that enables voice, data, and video collaboration over existing IP-based LANs, WANs, and the Internet. Specifically, IP Telephony uses open IETF and ITU standards to move multimedia traffic over any network that uses IP (the Internet Protocol). This offers users both flexibility in physical media (for example, POTS lines, ADSL, ISDN, leased lines, coaxial cable, satellite, and twisted pair) and flexibility of physical location. As a result, the same ubiquitous networks that carry Web, e-mail and data traffic can be used to connect to individuals, businesses, schools and governments worldwide. TAPI 3.0 is an evolutionary API that supports convergence of both traditional PSTN telephony and telephony over IP networks.

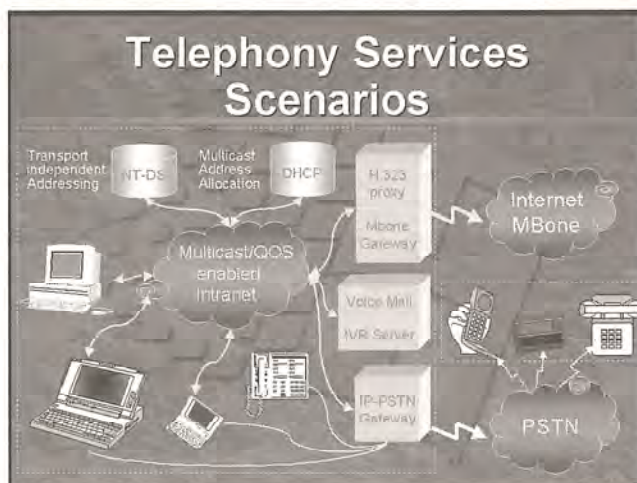
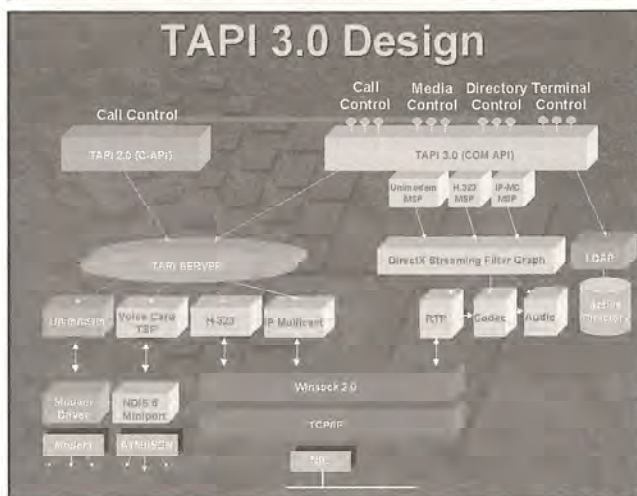
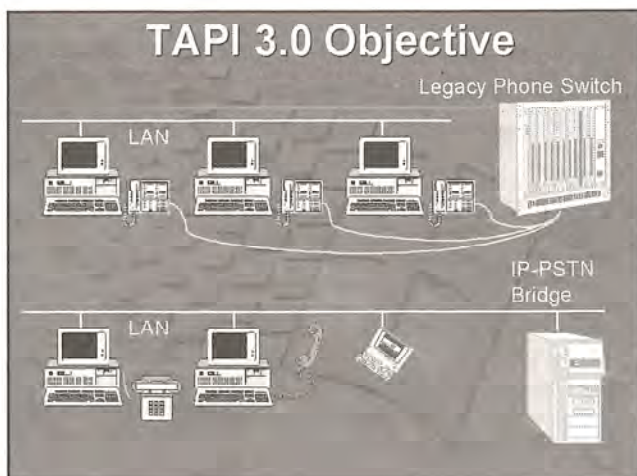
What are the Benefits of IP Telephony? IP Telephony allows organizations and individuals to lower the costs of existing services, such as voice and broadcast video, while at the same time broadening their means of communication to include modern video conferencing, application sharing, and whiteboarding tools. In the past, organizations have deployed separate networks to handle traditional voice, data, and video traffic. Each with different transport requirements, these networks were expensive to install, maintain, and reconfigure. Furthermore, since these networks were physically distinct, integration was difficult if not impossible, limiting their potential usefulness. IP Telephony blends voice, video and data by specifying a common transport, IP, for each, effectively collapsing three networks into one. The result is increased manageability, lower support costs, a new breed of collaboration tools, and increased productivity. Possible applications for IP Telephony include telecommuting, real-time document collaboration, distance learning, employee training, video conferencing, video mail, and video on demand.

What is TAPI 3.0? As telephony and call control become more common in the desktop computer, a general telephony interface is needed to enable applications to access all the telephony options available on any machine. Additionally, it is imperative that the media or data on a call is available to applications in a standard manner. TAPI 3.0 is an architecture that provides simple and generic methods for making connections between two or more machines, and accessing any media streams involved in that connection. It abstracts call-control functionality to allow different, and seemingly incompatible, communication protocols to expose a common interface to applications. IP Telephony is a demand poised for explosive growth, as organizations begin an historic shift from expensive and inflexible circuit-switched public telephone networks to intelligent, flexible and inexpensive IP networks.

Inside TAPI 3.0: TAPI 3.0 integrates multimedia stream control with legacy telephony.

T

TAPI 3.0 / TAPI 3.0



Additionally, it is an evolution of the TAPI 2.1 API to the COM model, allowing TAPI applications to be written in any language, such as JavaT, C/C++ and the Microsoft Visual Basic programming system. Besides supporting classic telephony providers, TAPI 3.0 supports standard H.323 conferencing and IP multicast conferencing. TAPI 3.0 utilizes the Windows NT 5.0 Active Directory service to simplify deployment within an organization, and it supports quality of service (QoS) features to improve conference quality and network manageability.

There are four major components to TAPI 3.0:

1. TAPI 3.0 COM API
2. TAPI Server
3. Telephony Service Providers
4. Media Stream Providers

In contrast to TAPI 2.1, the TAPI 3.0 API is implemented as a suite of Component Object Model (COM) objects. Moving TAPI to the object-oriented COM model allows component upgrades of TAPI features. It also allows developers to write TAPI-enabled applications in any language, such as Java, Visual Basic, or C/C++. The TAPI Server process (TAPISRV.EXE) abstracts the TSPI (TAPI Service Provider Interface) from TAPI 3.0 and TAPI 2.1, allowing TAPI 2.1 Telephony Service Providers to be used with TAPI 3.0, maintaining the internal state of TAPI. Telephony Service Providers (TSPs) are responsible for resolving the protocol-independent call model of TAPI into protocol-specific call control mechanisms. TAPI 3.0 provides backward compatibility with TAPI 2.1 TSPs. Two IP Telephony service providers (and their associated MSPs) ship by default with TAPI 3.0: the H.323 TSP and the IP Multicast Conferencing TSP. TAPI 3.0 provides a uniform way to access the media streams in a call, supporting the DirectShow API as the primary media stream handler. TAPI Media Stream Providers (MSPs) implement DirectShow interfaces for a particular TSP and are required for any telephony service that makes use of DirectShow streaming. Generic streams are handled by the application.

There are five objects in the TAPI 3.0 API:

1. TAPI
2. Address
3. Terminal
4. Call
5. CallHub

The TAPI object is the application's entry point to TAPI 3.0. This object represents all telephony resources to which the local computer has access, allowing an application to enumerate all local and remote addresses. An Address object represents the origination or destination point for a call. Address capabilities, such as media and terminal support, can be retrieved from this object. An application can wait for a call on an Address object, or can create an outgoing call object from an Address object. A Terminal object represents the sink, or renderer, at the termination or origination point of a connection. The Terminal object can map to hardware used for human interaction, such as a telephone or microphone, but can also be a file or any other device capable of receiving input or creating output. The Call object represents an address' connection between the local address and one or more other addresses (This connection can be made directly or through a CallHub). The Call object can be imagined as a first-party view of a telephone call. All call control is done through the Call object. There is a call object for each member of a CallHub. The CallHub object represents a set of related calls. A CallHub object cannot be created directly by an application. They are created indirectly when incoming calls are received through TAPI 3.0. Using a CallHub object, a user can enumerate the other participants in a call or conference, and possibly (because of the location independent nature of COM) perform call control on the remote Call objects associated with those users, subject to sufficient permissions:

Using TAPI Objects to Place a Call:

1. Create and initialize a TAPI object.
2. Use the TAPI object to enumerate all available Address objects on a machine (for example, network cards, modems, and ISDN lines).
3. Enumerate the supported address types of each Address object (for example, a phone number, IP address, and so on).
4. Choose an Address object based on queries for support for appropriate media (audio, video, and so on) and address types.
5. Use the CreateCall() method of the Address object to create a Call object associated with a particular address.
6. Select appropriate Terminals on the Call object.
7. Call the Connect() method of the Call object to place the call.

To Answer a Call:

1. Create and initialize a TAPI object.
2. Use the TAPI object to enumerate all available Address objects on a machine (for example, network cards, modems, and ISDN lines).
3. Enumerate the supported address types of each Address object (e.g. a phone number, IP address, etc.).
4. Choose an Address object based on queries for support of appropriate media (audio,

video, and so on) and address types.

5. Register an interest in specific media types with the appropriate Address object.

6. Register a call event handler (i.e. implement an `ITCallNotification` interface) with the Address object.

7. TAPI notifies the application of a new call through `ITCallNotification` and creates a Call object.

8. Select appropriate Terminals on the Call object.

9. Call the `Connect()` method of the Call object to place the call.

10. Call the `Answer()` method of the Call object to answer the call.

Media Streaming Model: The Windows operating system provides an extensible framework for efficient control and manipulation of streaming media called the DirectShow API. DirectShow, through its exposed COM interfaces, provides TAPI 3.0 with unified stream control. At the heart of the DirectShow services is a modular system of pluggable components called filters, arranged in a configuration called a filter graph. A component called the filter graph manager oversees the connection of these filters and controls the stream's data flow. Each filter's capabilities are described by a number of special COM interfaces called pins. Each pin instance can consume or produce streaming data, such as digital audio. While COM objects are usually exposed in user mode programs, the DirectShow streaming architecture includes an extension to the Windows driver model that allows the connection of media streams directly at the device driver level.

These high-performance streaming extensions to the Windows driver model avoid user-to-kernel mode transitions, and allow efficient routing of data streams between different hardware components at the device driver level. Each kernel mode filter is mirrored by a corresponding user mode proxy that facilitates connection setup and can be used to control hardware-specific features. DirectShow network filters extend the streaming architecture to machines connected on an IP network. The Real-Time Transport protocol (RTP), designed to carry real-time data over connectionless networks, transports TAPI media streams and provides appropriate time stamp information. TAPI 3.0 includes a kernel mode RTP network filter. TAPI 3.0 utilizes this technology to present a unified access method for the media streams in multimedia calls. Applications can route these streams by manipulating corresponding filter graphs; they can also easily connect streams from multiple calls for bridging and conferencing capabilities.

What is H.323? H.323 is a comprehensive International Telecommunications Union (ITU) standard for multimedia communications (voice, video, and data) over connectionless networks that do not provide a guaranteed quality of service, such as IP-based networks and the Internet. It provides for call control, multimedia management, and bandwidth management for point-to-point and multipoint conferences. H.323 mandates support for standard audio and video codecs and supports data sharing via the T.120 standard. Furthermore, the H.323 standard is network, platform, and application independent, allowing any H.323 compliant terminal to interoperate with any other.

H.323 allows multimedia streaming over current packet-switched networks. To counter the effects of LAN latency, H.323 uses as a transport the Real-time Transport Protocol (RTP), an IETF standard designed to handle the requirements of streaming real-time audio and video over the Internet. The H.323 standard specifies three command and control protocols:

1. H.245 for call control
2. Q.931 for call signaling
3. The RAS (Registration, Admissions, and Status) signaling function

The H.245 control channel is responsible for control messages governing operation of the H.323 terminal, including capability exchanges, commands, and indications. Q.931 is used to set up a connection between two terminals, while RAS governs registration, admission, and bandwidth functions between endpoints and gatekeepers (RAS is not used if a gatekeeper is not present).

H.323 defines four major components for an H.323-based communications system:

1. Terminals
2. Gateways
3. Gatekeepers
4. Multipoint Control Units (MCUs)

Terminals are the client endpoints on the network. All terminals must support voice communications; video and data support is optional.

A Gateway is an optional element in an H.323 conference. Gateways bridge H.323 conferences to other networks, communications protocols, and multimedia formats. Gateways are not required if connections to other networks or non-H.323 compliant terminals are not needed. Gatekeepers perform two important functions which help maintain the robustness

of the network - address translation and bandwidth management. Gatekeepers map LAN aliases to IP addresses and provide address lookups when needed. Gatekeepers also exercise call control functions to limit the number of H.323 connections, and the total bandwidth used by these connections, in an H.323 "zone." A Gatekeeper is not required in an H.323 system-however, if a Gatekeeper is present, terminals must make use of its services.

Multipoint Control Units (MCU) support conferences between three or more endpoints. An MCU consists of a required Multipoint Controller (MC) and zero or more Multipoint Processors (MPs). The MC performs H.245 negotiations between all terminals to determine common audio and video processing capabilities, while the Multipoint Processor (MP) routes audio, video, and data streams between terminal endpoints.

Any H.323 client is guaranteed to support the following standards: H.261 and G.711. H.261 is an ITU-standard video codec designed to transmit compressed video at a rate of 64 Kbps and at a resolution of 176x44 pixels (QCIF). G.711 is an ITU-standard audio codec designed to transmit A-law and μ -law PCM audio at bit rates of 48, 56, and 64 Kbps. Optionally, an H.323 client may support additional codecs: H.263 and G.723. H.263 is an ITU-standard video codec based on and compatible with H.261. It offers improved compression over H.261 and transmits video at a resolution of 176 x 44 pixels (QCIF). G.723 is an ITU-standard audio codec designed to operate at very low bit rates.

The H.323 Telephony Service Provider (along with its associated Media Stream Provider) allows TAPI-enabled applications to engage in multimedia sessions with any H.323-compliant terminal on the local area network. Specifically, the H.323 Telephony Service Provider (TSP) implements the H.323 signaling stack. The TSP accepts a number of different address formats, including name, machine name, and e-mail address. The H.323 MSP is responsible for constructing the DirectShow filter graph for an H.323 connection (including the RTP, RTP payload handler, codec, sink, and renderer filters).

H.323 telephony is complicated by the reality that a user's network address (in this case, a user's IP address) is highly volatile and cannot be counted on to remain unchanged between H.323 sessions. The TAPI H.323 TSP uses the services of the Windows NT Active Directory to perform user-to-IP address resolution. Specifically, user-to-IP mapping information is stored and continually refreshed using the Internet Locator Service (ILS) Dynamic Directory, a real-time server component of the Active Directory.

The following user scenario illustrates IP address resolution in the H.323 TSP:

1. John wishes to initiate an H.323 conference with Alice, another user on the LAN. Once Alice's video conferencing application creates an Address object and puts it in listen mode, Alice's IP address is added to the Windows NT Active Directory by the H.323 TSP. This information has a finite time to live (TTL) and is refreshed at regular intervals via the Lightweight Directory Access Protocol (LDAP).
2. John's H.323 TSP then queries the ILS Dynamic Directory server for Alice's IP address. Specifically, John queries for any and all RTPPerson objects in the Directory associated with Alice.

3. Armed with Alice's up-to-date IP address, John initiates an H.323 call to Alice's machine, and H.323-standard negotiations and media selection occurs between the peer TSPs on both machines. Once capability negotiations have been completed, both H.323 Media Stream Providers (MSPs) construct appropriate DirectShow filter graphs, and all media streams are passed off to DirectShow to handle. The conference then begins.

What is IP Multicast Conferencing? IP Multicast is an extension to IP that allows for efficient group communication. IP Multicast arose out of the need for a lightweight, scalable conferencing solution that solved the problems associated with real-time traffic over a datagram, "best-effort" network. There are many advantages to using IP Multicast: scalability, fault tolerance, robustness, and ease of setup. The IP Multicast conferencing model incorporates the following key features:

1. No global coordination is needed to add and remove members from a conference.
2. To reach a multicast group, a user sends data to a single multicast IP address. No knowledge of the other users in a group is necessary.
3. To receive data, users register their interest in a particular multicast IP address with a multicast aware router. No knowledge of the other users in a group is necessary. Routers hide the multicast implementation details from the user.

Traditional connection-oriented conferencing suffers from a number of problems:

1. User complexity: Users must know the location of every user they wish to converse with, limiting scalability and fault-tolerance and rendering it difficult for users to add and remove themselves from a conference.
2. Wasted bandwidth: A user wishing to broadcast data to N users must send data through

N connections.

The total bandwidth required for multiparty conferences in which all users are sending data goes up as N squared the number of parties involved, leading to huge scalability problems. IP Multicast takes advantage of the actual network topology to eliminate the transmission of redundant data down the same communications links.

IP Multicast implements a lightweight, session-based communications model, which places relatively little burden on conference users. Using IP Multicast, users send only one copy of their information to a group IP address that reaches all recipients. IP Multicast is designed to scale well as the number of participants expands. Adding one more user does not add a corresponding amount of bandwidth. Multicasting also results in a greatly reduced load on the sending server. IP Multicast routes these one-to-many data streams efficiently by constructing a spanning tree, in which there is only one path from one router to any other. Copies of the stream are made only when paths diverge.

Without multicasting, the same information must either be carried over the network multiple times, one time for each recipient, or broadcast to everyone on the network, consuming unnecessary bandwidth and processing. IP Multicast uses Class D Internet Protocol addresses to specify multicast host groups, ranging from 224.0.0.0 to 239.255.255.255. Both permanent and temporary group addresses are supported. Permanent addresses are assigned by the Internet Assigned Numbers Authority (IANA) and include 224.0.0.1, the "all-hosts group" used to address all multicast hosts on the local network, and 224.0.0.2, which addresses all routers on a LAN. The range of addresses between 224.0.0.0 and 224.0.0.255 is reserved for routing and other low-level network protocols. Other addresses and ranges have been reserved for applications, such as 224.0.13.000 to 224.0.13.255 for Net News (for more information, see RFC 1700, "Assigned Numbers" at <http://ftp.internic.net/rfc/rfc1700.txt>).

The transport protocol for IP Multicast is RTP (Real-time Transport Protocol), which provides a standard multimedia header giving timestamp, sequence numbering, and payload format information. Applications for IP Multicast include video and audio conferencing, telecommuting, database and Web-site replication, distance learning, dissemination of stock quotes, and collaborative computing. At present, the largest demonstration of the capabilities of IP Multicast is the Internet MBONE (Multicast Backbone). The MBONE is an experimental, global multicast network layered on top of the physical Internet. It has been in existence for about five years, and presently carries IETF meetings, NASA space shuttle launches, music, concerts, and many other live meetings and performances. www.mbone.com.

The IP Multicast Conferencing TSP is chiefly responsible for resolving conference names to IP multicast addresses, using the Session Description Protocol (SDP) conference descriptors stored in the ILS Dynamic Directory Conference Server. It is complemented by the Rendezvous conference controls, described later. The IP Multicast Conferencing MSP is responsible for constructing an appropriate DirectShow filter graph for an IP multicast connection (including RTP, RTP payload handler, codec, sink, and renderer filters).

TAPI 3.0 uses the IETF standard Session Description Protocol to advertise IP multicast conferences across the enterprise. SDP descriptors are stored in the Windows NT Active Directory — specifically, in the ILS Dynamic Directory Conference Server. In contrast to the Dynamic Directory servers used by the H.323 TSP, there is only one ILS Conference Server per enterprise, since conference announcements are not continually refreshed, therefore consuming little bandwidth.

TAPI 3.0's IP multicast conference mechanism is illustrated in the following scenario, in which John wishes to initiate a multicast conference:

1. John's TAPI 3.0-enabled application uses the Rendezvous Controls to create an SDP session descriptor on the ILS Conference Server. The SDP descriptor contains, among other things, the conference name, start and end time information, the IP multicast address of the conference, and the media types used for the conference.
2. Jim queries the ILS Conference Server for SDP descriptors of conferences matching his criteria.
3. Mary and Alice perform similar queries and use the SDP information they receive to decide to participate in John's conference. Armed with the multicast IP address of the conference, they join the multicast host group.

The Rendezvous Controls are a set of COM components that abstract the concept of a conference directory, providing a mechanism to advertise new multicast conferences and to discover existing ones. They provide a common schema (SDP) for conference announcement, as well as scriptable interfaces, authentication, encryption, and access control features.

The user may add, delete, and enumerate multicast conferences stored on an ILS

Conference Server via the Rendezvous Controls. These controls manipulate conference data via the Lightweight Directory Access Protocol (LDAP). The conferencing application uses the Rendezvous Controls to obtain session descriptors for the conferences that match the user's criteria. Access control lists (ACLs) protect each of the stored conference announcements, and whether or not an announcement is visible and accessible depends upon the user's credentials. Once the user has chosen a conference, the user application searches for all Address objects that support the address type "Multicast Conference Name." The application then uses the conference name from the SDP descriptor as a parameter to the CreateCall() method of the appropriate Address object, passes the appropriate Terminal objects to the returned Call object, and calls Call->Connect(). The Rendezvous Controls store the conference information on an ILS Conference Server in a format defined by the Session Description Protocol (SDP), an IETF standard for announcing multimedia conferences. The purpose of SDP is to publicize sufficient information about a conference (time, media, and location information) to allow prospective users to participate if they so choose. Originally designed to operate over the Internet MBONE (IP Multicast Backbone), SDP has been integrated by TAPI 3.0 with the Windows NT Active Directory, thereby extending its functionality to local area networks. An SDP descriptor advertises the following information about a conference.

A session description is broken into three main parts: a single Session Description, zero or more Time Descriptions, and zero or more Media Descriptions. The Session Description contains global attributes that apply to the whole conference or all media streams. Time Descriptions contain conference start, stop, and repeat time information, while Media Descriptions contain details that are specific to a particular media stream.

While traditional IP multicast conferences operating over the MBONE have advertised conferences using a push model based on the Session Announcement Protocol (SAP), TAPI 3.0 utilizes a pull-based approach using Windows NT Active Directory services. This approach offers numerous advantages, among them bandwidth conservation and ease of administration.

TAPI 3.0's conference security system addresses the following needs:

- * Controlling who can create, delete, and view conference announcements.
- * Preventing conference eavesdropping.

TAPI 3.0 utilizes the security features of the Windows NT Active Directory and LDAP to provide for secure conferencing over insecure networks such as the Internet. Each object in the Active Directory can be associated with an Access Control List (ACL) specifying object access rights on a user or group basis. By associating ACLs with SDP conference descriptors, conference creators can specify who can enumerate and view conference announcements. User authentication is provided by the Windows NT security subsystem.

Session Descriptors are transmitted from the ILS Conference Server to the user over LDAP in encrypted form, via a Secure Sockets Layer (SSL) connection, ensuring that the SDP is safe from eavesdroppers. IP Multicast makes no provision for authenticating users. Any user may anonymously join a multicast host group. To keep conferences private, TAPI 3.0 allows an IP multicast conference to be encrypted, with the encryption key distributed from within the conference descriptor. Only users with sufficient permissions have access to a conference's SDP descriptor, and therefore the Multicast Encryption Key. Once an authenticated user fetches the encryption key, he or she can participate in the conference.

In contrast to traditional data traffic, multimedia streams, such as those used in IP Telephony or videoconferencing, may be extremely bandwidth and delay sensitive, imposing unique quality of service (QoS) demands on the underlying networks that carry them. Unfortunately, IP, with a connectionless, "best-effort" delivery model, does not guarantee delivery of packets in order, in a timely manner, or at all. In order to deploy real-time applications over IP networks with an acceptable level of quality, certain bandwidth, latency, and jitter requirements must be guaranteed, and must be met in a fashion that allows multimedia traffic to coexist with traditional data traffic on the same network.

Bandwidth: Multimedia data, and in particular video, may require orders of more bandwidth than traditional networks have been provisioned to handle. An uncompressed NTSC video stream, for example, can require upwards of 220 megabits per second to transmit. Even compressed, a handful of multimedia streams can completely overwhelm any other traffic on the network.

Latency: The amount of time a multimedia packet takes to get from the source to the destination (latency) has a major impact on the perceived quality of the call. There are many contributors towards latency, including transmission delays, queuing delays in network equipment, and delays in host protocol stacks. Latency must be minimized in order to maintain a certain level of interactivity and to avoid unnatural pauses in conversation.

Jitter: In contrast to data traffic, real-time multimedia packets must arrive in order and on time to be of any use to the receiver. Variations in packet arrival time (jitter) must be below a certain threshold to avoid dropped packets (and therefore irritating shrieks and gaps in the call). Jitter, by determining receive buffer sizes, also affects latency.

Coexistence: In comparison with multimedia traffic, data traffic is relatively bursty, and arrives in unpredictable chunks (for instance, when someone opens a Web page, or downloads a file from an FTP site). Aggregations of such bursts can clog routers and cause gaps in multimedia conferences, leaving calls at the mercy of everyone on the network (including other IP Telephony users). Multimedia bandwidth must be protected from data traffic, and vice versa.

Public-switched telephone networks guarantee a minimum quality of service by allocating static circuits for every telephone call. Such an approach is simple to implement, but wastes bandwidth, lacks robustness, and makes voice, video, and data integration difficult. Furthermore, circuit-switched data paths are impossible to create using a connectionless network such as IP.

QoS support on IP networks offers the following benefits:

1. Support for real-time multimedia applications.
2. Assurance of timely transfers of large amounts of data.
3. The ability to share the network in a manner that avoids starving applications of bandwidth.

Quality of service in TAPI 3.0 is handled through the DirectShow RTP filter, which negotiates bandwidth capabilities with the network based on the requirements of the DirectShow codecs associated with a particular media stream. These requirements are indicated to the RTP filter by the codecs via its own QoS interface. The RTP filter then uses the COM Winsock2 QoS interfaces to indicate, in an abstract form, its QoS requirements to the Winsock2 QoS service provider (QoS SP). The QoS SP, in turn, invokes a number of varying QoS mechanisms appropriate for the application, the underlying media, and the network, in order to guarantee appropriate end-to-end QoS. These mechanisms include:

- a. The Resource Reservation Protocol (RSVP)
- b. Local Traffic Control: Packet Scheduling; 802.1p; Appropriate layer 2 signaling mechanisms
- c. IP Type of Service and DTR header settings

The Resource Reservation Protocol (RSVP) is an IETF standard designed to support resource (for example, bandwidth) reservations through networks of varying topologies and media. Through RSVP, a user's quality of service requests are propagated to all routers along the data path, allowing the network to reconfigure itself (at all network levels) to meet the desired level of service. The RSVP protocol engages network resources by establishing flows throughout the network. A flow is a network path associated with one or more senders, one or more receivers, and a certain quality of service. A sending host wishing to send data that requires a certain QoS will broadcast, via an RSVP-enabled Winsock Service Provider, "path" messages toward the intended recipients. These path messages, which describe the bandwidth requirements and relevant parameters of the data to be sent, are propagated to all intermediate routers along the path. A receiving host, interested in this particular data, will confirm the flow (and the network path) by sending "reserve" messages through the network, describing the bandwidth characteristics of data it wishes to receive from the sender. As these reserve messages propagate back toward the sender, intermediate routers, based on bandwidth capacity, decide whether or not to accept the proposed reservation and commit resources. If an affirmative decision is made, the resources are committed and reserve messages are propagated to the next hop on the path from source to destination.

Packet Scheduling: This mechanism can be used in conjunction with RSVP (if the underlying network is RSVP-enabled) or without RSVP. Traffic is identified as belonging to one flow or another, and packets from each flow are scheduled in accordance with the traffic control parameters for the flow. These parameters generally include a scheduled rate (token bucket parameter) and some indication of priority. The former is used to pace the transmission of packets to the network. The latter is used to determine the order in which packets should be submitted to the network when congestion occurs.

802.1p: Traffic control can also be used to determine the 802.1 User Priority value (a MAC header field used to indicate relative packet priority) to be associated with each transmitted packet. 802.1p-enabled switches can then give preferential treatment to certain packets over others, providing additional quality of service support at the data link layer level.

Layer 2 Signaling Mechanisms: In response to Winsock 2 QoS APIs, the QoS service provider may invoke additional traffic control mechanisms depending on the specific under-

lying data link layer. It may signal an underlying ATM network, for instance, to set up an appropriate virtual circuit for each flow. When the underlying media is a traditional 802 shared media network, the QoS service provider may extend the standard RSVP mechanism to signal a Subnet Bandwidth Manager (SBM). The SBM provides centralized bandwidth management on shared networks.

Each IP packet contains a three-bit Precedence field, which indicates the priority of the packet. An additional field can be used to indicate a delay, throughput, or reliability preference to the network. Local traffic control can be used to set these bits in the IP headers of packets on particular flows. As a result, packets belonging to a flow will be treated appropriately later by three devices on the network. These fields are analogous to 802.1p priority settings but are interpreted by higher layer network devices. www.microsoft.com/communications. See H.323.

TAPTS Tenant Package.

TAR TAR is the UNIX standard program for combining and compressing files. It's like a UNIX Winzip program. I believe it stands for "Tape Archiver". The UNIX System V manual lists the command description as "tape file archiver". One no longer uses it just for tape archiving, but that was its initial usage.

TARFU TARFU is a military term used on TTY circuits to state that "things are really F...ked up". In the military there were three stages to a project, which is not going well — SNAFU, FUBAR and TARFU. See SNAFU and FUBAR.

TARGA Truevision Advanced Raster Graphics Adapter.

Target A SCSI device that performs an operation requested by an initiator.

Target Host Number The number that identifies the destination software program during the user logon process.

Target ID Address Resolution Protocol. TARP is the established multi-vendor standard for SONET network elements that support TLI Operating System interfaces.

Target Token Rotation Time TRT. An FDDI (Fiber Distributed Data Interface) token travels along the network ring from node to node. If a node does not need to transmit data, it picks up the token and sends it to the next node. If the node possessing the token does need to transmit, it can send as many frames as desired for a fixed amount of time.

Tariff Documents filed by a regulated telephone company with a state public utility commission or the Federal Communications Commission. The tariff, a public document, details services, equipment and pricing offered by the telephone company (a common carrier) to all potential customers. Being a "common carrier" means it (the phone company) must offer its services to everybody at the prices and at the conditions outlined in its public tariffs. Tariffs do not carry the weight of law behind them. If you or the telephone company violate them, no one will go to jail. The worst that can happen to you, as a subscriber, is that your service will be cut off, or threatened to be cut off. Regulatory authorities do not normally approve tariffs. They accept them — until they are successfully challenged before a hearing of the regulatory body or in court (usually Federal Court). Many tariffs were accepted by regulatory commissions only to be struck down in court as unlawful, discriminatory, not cost-justified, etc. Monies collected under the tariff have been refunded and unnecessary equipment removed. In these new, competitive days, many telephone companies are violating their own tariffs by charging less money than their tariffs say they should, or bundling services together at a discount. They are also providing service and equipment on terms less onerous than outlined in their tariffs. Many users now regard tariffs as starting bargaining points, rather than ending bargaining points.

Tariff 12 A user-specific long distance tariff of AT&T. Tariff 12 gives AT&T the ability to price its long distance services for one company practically any way it feels — giving them a mix of services at stable prices over the long term with significant volume discounts. As this dictionary was going to the printer, a federal appeals court overturned the Federal Communications Commission's April 1989 decision allowing AT&T to offer custom networks and ordered the FCC to reopen its investigation into the legality of the Tariff 12 deals. There are still some users. They are "grandfathered" until we get a final say on the tariff. And, as we go to press, AT&T can offer Tariff 12 customized services to any company — but cannot include 800 services in its Tariff 12 pricing.

Tariff 15 A user-specific long distance tariff of AT&T. Tariff 15 gives AT&T the ability to price its long distance services for one company practically any way it feels. Tariff 15 is single-customer discounting. Some of AT&T competitors claim the tariff is "illegal."

Tariff Rebalancing Largely an initiative of the FCC in the US, national, regional (e.g., EU) and international (ITU) regulatory authorities and policy-making bodies are con-

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Harry Newton has 31 years studying, writing, lecturing and investing in telecommunications. He founded LAN (now Network), Teleconnect, Call Center and Computer Telephony magazines. He also founded the enormously successful trade shows, Computer Telephony Conference and Exposition (CT Expo) and Call Center Demo. Newton has an MBA from the Harvard Business School and an Economics degree from the University of Sydney, Australia.

He is not an engineer. He wrote this dictionary to learn enough to be dangerous in front of engineers. That, he says, is the ultimate thrill. Newton explains complex technology in non-technical business language even he can understand. He believes (perhaps naively) that anyone in business should be able to understand complex technical terms — whatever it takes. As a result, some of his definitions are short. Some are long. He strives to be 100% explanatory and 100% practical — explaining what the technology means, what it does, what its benefits are and what its pitfalls are.

Most dictionaries are updated every ten years. Not this one. Newton and his staff update the dictionary every day and issue a new expanded, updated edition every year. This one is current as of early 2002.

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