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President's Corner

Mi Column es Su Column

This marks my inaugural column in PSR as TAPR President. Some of you may remember me from my past incarnation on the TAPR Board of Directors. It seems like another lifetime ago, but it wasn't so long ago that TAPR was born in Tucson. In fact, next year will mark the twenty-fifth anniversary of our incorporation.

To celebrate, we are taking the Digital Communications Conference (DCC) and the TAPR Board Meeting back to Tucson. We will also be making an attempt to reconnect with our roots while presenting a roadmap for where we are going with the group. Membership input will be solicited on both fronts.

Your board believes that we know where TAPR should go, but of course we would like some feedback from you to confirm this. We are moving to more widely embrace and promote cutting edge technology, and while this is not really new for us, it does mean that we will be contemplating a shift in focus so that packet will not really be our main or only focus. Digital modes and technology will still be central, but I think you'll appreciate our attempts to marry digital with analog (you remember analog, don't you?) and microwaves, and anything else that you might consider "bleeding edge". You'll be seeing more references to the VNA (Vector Network Analyzer), Reflock, and TADD. You'll be reading more about these projects in the next year. I hope that we'll be hearing about projects that you have been musing over, and perhaps we can help you make them a reality and bring them to fruition.

We believe that we need to revitalize our membership, and we will be presenting some ideas to enhance your membership experience and involvement in projects. We are continuing to streamline the management of the office, and to assist this initiative, I will draw your attention to the improvements that have been made to the web site. The responses we have seen so far have been overwhelmingly positive, and we welcome further feedback.

I would like to make this column YOUR column, not mine, so please feel free to contact me at **ve3gyq@tapr.org** with your questions and ideas, and I will try to address them directly and in future columns.

May you have a safe and happy Holiday Season.

73,

Dave VE3GYQ/W8

Spencerville, Ohio

###

TAPR Elects New President

TAPR announces the election of David Toth, MD, VE3GYQ, of Spencerville, Ohio as the corporation's new President, effective September 23, 2005. Dr. Toth succeeds John Ackermann, N8UR, of Dayton, Ohio.

"I've enjoyed serving as TAPR's President for the last five years, and we have accomplished a lot during that time," Ackermann said. "But five years is long enough for any organization to be led by one person, and I'm happy to turn the reins over to Dave Toth. Dave will bring new energy and ideas to TAPR, and I look forward to working with him in his new role." Ackermann will continue as a member of TAPR's Board of Directors.

Other TAPR officers Steve Bible, N7HPR (Vice President), Stan Horzepa, WA1LOU (Secretary), and Tom Holmes, N8ZM (Treasurer) were re-elected at the Board of Director's meeting held in Santa Ana, California, during this year's ARRL-TAPR Digital Communications Conference (DCC). Bible and Horzepa, as well as Darryl Smith, VK2TDS, were also re-elected by the membership to new three year terms as Directors.

Toth returned to the Board last year, after having served as a Director and Executive Vice President in the 1980s. "John Ackermann encouraged me to rejoin the board", Toth said. "What convinced me to re-engage was the path that he and the board had set TAPR on. Projects like the VNA (Vector Network Analyzer) were in the pipe, and the board was actively trying to redefine TAPR as a leader in Amateur Radio's technical advancement.

"We started in Tucson and pioneered in packet radio, but we are now a global organization with members around the globe, including a board member participating from Australia. We have broadened our horizons and are looking to refocus our efforts on advanced communications modes. I will be looking for input from the membership as to where we should be going and how we work to get there. And what gives me confidence about being successful in this role is the fact that we have a committed Board of Directors. Some folks might be nervous stepping back into a role likethis, but I am bolstered by the fact that I'll have Steve Bible as VP on my right, and John Ackermann as Past President on my left. And most importantly, we will be continuing the Board's work towards removing dependence on any one individual to guide the organization.

"Steve Bible continues to guide us through cutting-edge projects like the Reflock II. John Ackermann has been turning his attention to time- and frequency-related projects. John is a real time-keeping nut, and he's trying to infect the rest of us. John and Laura Koster continue to streamline our office management, and Tom Holmes has been working with them to not only manage, but also better understand, our finances. Stan Horzepa continues to turn out an excellent *Packet Status Register* every quarter. Darryl Smith has been playing with Google Earth and amazing us with its treasure trove of features. Other board members have been busy with projects and interests of their own.

"However, what interests me the most is what our membership has been doing. The presentations at the Digital Communications Conference clearly show the creativity that exists in our ranks. Where else but at the DCC would yoube able to hear presentations as diverse, yet challenging, as W3NRG's report on anomalous 10M propagation, AE5PL's proposal of a new method of routing AX.25 packets, and K7GNU's experimentation with passive radar. And those are only three of a dozen thought-provoking presentations that attendees heard.

"To build on this creative energy, we will be looking at innovative ways to expand and involve the membership. We want to create a fertile playing field where folks can interact and exchange ideas. We believe that this will be a way to develop our future projects more quickly and more efficiently. And of course, I encourage folks to write to me directly at **ve3gyq@tapr.org**."

###

2005 Digital Communications Conference Report

By Steven Bible, N7HPR, n7hpr@tapr.org

The 24th Annual ARRL and TAPR Digital Communications Conference was held at the Embassy Suites – Orange County Airport North in Santa Ana, California on September 23-25, 2005. Attendance was just over 100 people. A great time was had by all.

Friday and Saturday was filled with talks spanning a breadth of topics. Darryl Smith, VK2TDS, led off with a talk about Google Earth – Applications for Ham Radio. Ken Chong, WB6MLC, and Bill Prats, K6ACJ, spoke about using PSK63 for APRS HF Pack. Ed Sack, W3NRG, reported on the results of PropNet 10 Meter Experimentation. Mel Whitten, K0PFX, told all about WinDRM, an HF digital voice/data mode. John Ackermann, N8UR, spoke about TAPR's many timing products that are coming soon in kit form. Tom Holmes, N8ZM, and Tom McDermott, N5EG, talked about how vector network analyzers work and updated us on the TAPR VNA project. Jim McClellan, N5MIJ, talked about what the Texas group has been doing with D-Star. Finishing out the long list of talks was Software Defined Radio topics by Gerald Youngblood, K5SDR, Matt Ettus, N2MJI, and Eric Blossom, K7GNU.

Demonstration Room

The demonstration room was filled to capacity yet again. The Texas group brought their portable D-Star repeater setup and demonstrated its ability over the Internet. Matt and Eric showed off the USRP and GNU Radio. Gerald showed off the capabilities of the SDR-1000. This is only a fraction of the projects that were on display and demonstration.

2004 Doug DeMaw, W1FB, Technical Excellence Award

The highlight of the awards presentations at the banquet was the awarding of the 2004 Doug DeMaw, W1FB, Technical Excellence Award Cup to Tom McDermott, N5EG, and Karl Ireland for the QEX article "A Low-Cost 100 MHz Vector Network Analyzer with USB Interface" presented by Fried Heyn, WA6WZO, Honorary ARRL Vice-President.

Award Plaques were also given to Dan Cregg, KB6ENX, thanking him for being our Banquet Speaker, Tom and Karl for the TAPR/TenTec VNA project, Luis Cupido, CT1DMK thanking him for the Reflock II project, and John Bennett N4XI, thanking him for the series of weather station projects.

Many thanks to the donors of the banquet prize drawings:

- Sony GXB5005 GPS Evaluation Kit donated by Synergy Systems, LLC
- Insteon Starter Kit donated by Smarthome
- Two OpenTracker kits donated by Scott Miller, N1VG
- Three copies of Lighting Protection and Grounding Solutions for Communications Sites" donated by Rob Block, KB2UYT
- Three copies of Ubuntu Linux donated by Bdale Garbee, KB0G
- Three QEX Subscriptions donated by the ARRL
- 2006 ARRL Handbook donated by the ARRL
- Wireless Digital Communications: Design and Theory book donated by TAPR
- 20th, 21st, 22nd, and 23rd DCC Proceedings donated by TAPR
- Spread Spectrum Update book donated by TAPR

Next Year

Planning for next years DCC is already underway. The year 2006 will be the 25th Annual Digital Communications Conference and the 25th Anniversary of the incorporation of TAPR. Therefore, the 2006 DCC will be held in Tucson, Arizona for a homecoming of sorts. We hope you will join us next for a special DCC celebration.

###

New SIG for AX.25 Layer 2 Discussions

By Pete Loveall, AE5PL, pete@ae5pl.net

TAPR has implemented a new SIG for discussion of AX.25 as a Link Layer (Layer 2) protocol. Specifically, this SIG will be used to discuss implementations, current and future, of AX.25 which support higher level protocols as found in the OSI model (the same model currently used for networks throughout the world and what AX.25 was originally designed to support). Discussions will be focused on how different implementations affect reliability and usability of the RF network and how those areas can be improved.

You can subscribe by visiting **https://www.tapr.org/cgi-bin/mailman/listinfo/ax25-layer2.** I highly encourage all digipeater authors to participate as this will probably be the primary topic over the short term. I am posting on this on the APRSSIGs as UI digipeating will be one of the subjects discussed and will directly affect the future of APRS networks.

###

20th Annual SW Ohio Digital Symposium

By Hank Greeb, N8XX, hgreeb@one.net

The 20th Annual Southwest Ohio Digital Symposium will be held on 14 January 2006, Registration 08:00, technical sessions 09:00-1600 EST. Location is the Middletown Campus, Miami University, 4200 N University Blvd, Middletown, OH. This is the 2nd longest continuously running digital symposium in the United States, only surpassed by the TAPR/ARRL Digital Communications Conference.

We will discuss all types of digital modes, including 802.11b 2.4 GHz communications, use of digital modes in recent Hurricane relief efforts, experimentation with APRS, PSK-31, MFSK, etc. We typically accept papers until a week or so before the event, so if you have a pet mode or new experimental results of an existing mode, please contact Hank Greeb, **n8xx@arrl.net**. See **http://www.swohdigi.org/** for more details.

This symposium is free - no admission charge. Optional catered lunch at cost.

Talk-In Freq: 146.61, 224.96, and 444.825 with standard offsets.

###

AX.25 as a Layer 2 Protocol

By Pete Loveall, AE5PL, pete@ae5pl.net

For those of you who have followed my series of papers on the NSR digipeater algorithm, you have seen a transition from simply being APRS network fix to a focus on supporting non-APRS networks as well. TAPR has established a special interest group, which is focusing on revitalizing AX.25 as a link layer (layer 2) protocol. This paper looks at how the NSR algorithm has evolved into a generic digipeater algorithm and what part it plays in the overall scheme of utilizing AX.25 as a layer 2 protocol. It also looks at why migrating AX.25 networks to a generic layer 2 architecture is important to the Amateur Radio community.

AX.25 was developed during the late 1970s and early 1980s as the first non-RTTY digital protocol for Amateur Radio. It was loosely based on the various X.25 standards with certain adaptations for the broadcast, multipoint nature of radio. It also adopted and addressing schema utilizing the amateur's call sign and a numerical station identifier. The station identifier (SSID) allowed an amateur to have more than one station on the same AX.25 link. Another variance in the AX.25 specification is the allowance for digital repeaters to be identified in the protocol. This repeater identification was intended to prevent loops and to allow amateurs to explicitly denote best path. AX.25 was designed as a link layer (OSI/ISO layer 2) protocol, which could be used with higher layers, or as a standalone protocol.

For the most part, amateurs have implemented specialized AX.25 networks where there is either no higher layer protocol or where only a single layer 3 protocol can be carried. Amateurs have tried to make AX.25 into a layer 3 protocol even though it contains none of the fields necessary for such an implementation to be truly successful. The result has been networks which are single use, difficult to use, difficult to maintain, unreliable, and which fade into obscurity as other more robust networks become commonplace.

Amateur Radio is noted for providing dependable communications when commercial communications fail. This can be traced primarily to our ability to establish temporary communications infrastructures that do everyone can readily utilize. These infrastructures vary from CW traffic nets on HF to voice nets on VHF. Note that digital networks are sadly missing from this mission. Why? Because our forays into digital communications have been sadly focused on establishing these specialty networks without consideration for what is actually needed during an emergency or even during day-to-day "normal" communications.

The evolution of the NSR algorithm brought to light a key piece that had been missing from our AX.25 networks: the ability for layer 3 (and above) protocols to discover other layer 3 devices on the various links. It also brought to light something that had been written about back in the 1980s but ignored by our current network architects: multi-hop digipeating significantly degrades reliability and bandwidth. Both of these issues pointed to the need for reassessing our AX.25 networks and begin looking at them as individual links, not networks in of themselves.

The NSR algorithm has been modified to eliminate any dependencies on APRS. As such, it does not directly support any of the APRS routing protocols such as n-n or SSID routing. As can be seen at www.ax25.net, it is now a "pure" AX.25 digital repeater algorithm, which supports multi-use, simple to use, simple to maintain, reliable link layer communications. The key to this is that the link layer works whether there is a digipeater on the frequency or not. There is no pre-knowledge of the link architecture required of the individual devices attached to the link. They can now use generic discovery algorithms to ascertain best paths automatically. The digipeater operators define what constitutes the local link and rogue operators who would try to use excessive digipeater hops are prevented from degrading the local link.

The NSR algorithm can still perform its original mission, cleaning up the APRS channel, but now addresses a much broader issue: a generic multi-use link layer. But the NSR algorithm is just a small piece of the overall effort to produce a generic layer 2. The AX.25 Layer 2 SIG is focused on the AX.25 specification, inter-layer interfaces, and actual implementation issues. But the SIG is addressing all of these issues with one goal: for amateurs to have the ability to establish AX.25 digital data links anywhere, anytime that can be used by any properly licensed amateur with compatible equipment. To do this, a truly generic link layer with well-defined interfaces must be established. We hope to see this begin to occur over the next year.

###

Towards a Next Generation Amateur Radio Network

By Samuel A. Falvo II, KC5TJA/6, kc5tja@arrl.net

Abstract

The Next Generation Amateur Radio Network (NgARN) is a proposal to provide a logically and physically independent computer inter-network infrastructure for the Amateur Radio community. The intended mission profile of NgARN ranges from research and development applications to providing an emergency backup infrastructure for today's commercial Internet and Internet-2 systems.

About This Document

Where possible, I generally try to follow the MLA style guidelines. Due to the limitations of the document preparation system I used to write this text, however, there are a few typographical conventions, which the reader may not be familiar with. All bibliographic references follow IEEE-style citations (e.g., [1]), but MLA-inspired references. All references to endnotes appear as, e.g., [e5]. Elements inside single asterisks appear *in italics.*

This document was prepared using ReStructured Text tools (**http://docutils.sf.net**).

Introduction

In modern society, the Internet has become truly ubiquitous. Web URLs are seemingly everywhere in television and radio advertisements. Newspapers and other traditional print media have adapted to the Internet. Research tools' including Google and LexisNexis provides content that is otherwise nearly impossible for laymen to get.

For others, the Internet provides an effective replacement for traditional communications technologies. Many businesses are adopting Voice-over-IP for their internal, and sometimes even external, communications needs. Many people rely on e-mail for their primary communications channel, choosing to rely on a cell phone only for more time-sensitive communications. I even lack a TV or any newspaper deliveries, opting to read the equivalent material via commonly available websites [e1]_. Clearly, the Internet has become a staple personal, political, academic, and commercial infrastructure in today's society.

For as great as the Internet is, it builds upon a fabric that has many opportunities for failure. With the commercial telecommunications providers in control of the underlying physical media, it has no backup infrastructure in the event of a widespread emergency. Should a copper or glass connection be severed, or power go out, the affected parties and *all down-stream nodes* are flat out of luck until that connection is replaced. [e2]_ Even many wireless ISPs use wireless only in a "last mile" application, still requiring cabled backbone connections from their cell towers.

Another critical problem with the current Internet is the scarcity of IP addresses. NAT, or Network Address Translation, techniques are often used to help mitigate this problem [1]_, but are only delaying the inevitable. Even with the existence of NAT, the IP scarcity was sufficient to cause Stanford University to relinquish its previously long-held class-A address range as early as 1998, in favor of 3 class-B address blocks [3]_. Still, naysayers continue to claim the contrary [4]_, backed by evidence of their own, of course.

When purchasing dedicated servers, for example, it is often true that it receives one and only one IP address by default, encouraging such tricks as name-based virtual hosting. While there is nothing innately wrong with name-based virtual hosting, many spam-filtering techniques depend on accurate reverse name look-ups to properly "authenticate" a domain for sending mail. Since only one domain name can be mapped back from any single IP, it's easy to see how this mechanism can break down with name-based virtual hosting. This has led to alternative, far less effective approaches, such as SPF, POP-before-relay, etc.

Many dedicated server customers are upset over this, as it often breaks their branding efforts. Compound this with the necessity for SSL-certified websites to have a dedicated IP when it otherwise doesn't need one, and it's easy to see how there can be many strange interactions and hidden costs involved with hosting multiple websites with an ISP. Customers who do not understand the problems leading to their IP provisioning policies often wrongly accuse many ISPs for shady billing practices.

Another reason why NAT is detrimental to the Internet is that it often adds unnecessary delay to time-critical applications. In order to work around NAT barriers, for example, many applications that normally use UDP for time-sensitive packet delivery (e.g., games, voice-over-IP, etc.) are forced to employ a work-around which requires the use of a TCP channel. Since NAT *requires* virtual circuits to identify paths through a network, and since TCP establishes a "reliable" virtual circuit connection, it follows that it is a quick and easy work-around to the NAT barrier. However, TCP's reliability results in significant packet jitter, often resulting in many detrimental effects. [e3]_

The last flaw I'll choose to list here is spam. Spam often accounts for up to 45% of a network's global traffic [5]_. Spam is quite definitely a problem, which stems from the relative insecurity of SMTP-based e-mail. However, commercial ISPs are unwilling to switch to more secure and better-architected alternatives, fearing lack of adoption, and therefore irretrievable loss of research revenue.

All of these issues presents a unique opportunity for the Amateur Radio service as a whole: to arrive at a next generation internetwork that is logically independent as well as physically independent from the current commercial Internet. Because the amateur service isn't bound by commercial interests looking to maintain compatibility with existing infrastructures, we needn't be hindered with past technologies, presenting a unique opportunity for Amateur Radio operators to once again be on *the* cutting edge, thus fulfilling one of the legal requirements justifying the Amateur Radio service.

As with all good-for-the-public ideas, there is a selfish motive for proposing a new wireless-only Internet as well. Participating in such a network is fun, challenging, and quite educational, on multiple levels. All my life I've always wanted to play with ATM, Internet, and other communications technologies. While I get to play with Internet technologies all the time at work now, ATM, token ring, and other layer-2 technologies are still well outside my reach. NgARN gives me, and many others, the opportunity to play with these technologies, and apply them where they are appropriate, while still providing a consistent internetwork architecture. Additionally, unlike paid Internet services, once the equipment is paid for, the service itself is essentially free.

It also holds the potential to retain a larger amount of amateurs in the service as well as to draw new members. I once heard a conversation on the local 2m repeater discussing the problem of kids not being interested in ham radio. Many ideas were discussed for making it appealing to them. "All they seem interested in is their stupid instant messengers," reported the disgruntled proponent, whose call sign I've deliberately withheld for privacy reasons. Never was packet radio mentioned. What is instant messaging, if not a keyboard-to-keyboard application of packet technology, precisely the same technology that some people in the amateur service formerly said packet radio was *unsuitable* for? [6]_

Requirements for NgARN

In order for the NgARN to be successful, it must fulfill certain requirements, covering a wide spectrum.

* It must be affordable,

* It must be easy to deploy,

* It must advance the current state of the art *somehow*, and,

* It must provide an excellent return on investment.

Affordability

It is vital that using NgARN must be cost-competitive with commercial Internet services in order to be appealing to the discriminating amateur, especially where customized hardware is required.

To provide a representative example for comparison, I picked Verizon DSL as a "competitor" with whom NgARN must compete against [e4]_. They are not the cheapest, but not the most expensive either. They also have a nationwide presence, which makes them a good target for comparison, at least in the United States.

According to [7]_, you can obtain DSL for $19.95/month for the first three months, and $29.95/month for the remaining months, assuming you purchase a full year in advance. That totals $329.40 for their one-year commitment. What they're unwilling to tell you on their website is that it is only 768Kbps downstream, 128Kbps upstream. Assuming you maximize both sub-channels, that's a total of 896Kbps -- not even a full Mbps link. This drives the cost per Mbps to $367.63/Mbps/yr. Note the "per-year" unit -- the customer is expected to pay at least that amount every year [e5]_.

Amateur equipment must be cost-competitive with this price point in order to be accepted into the amateur service. Current TNC offerings commonly found offer at best 9600bps links using G3RUH modulation techniques. According to [8]_, the cost for a PK232 9600bps multimode TNC can be as high as $519.95, for a cost per Mbps of $54,161.46/Mbps!

Fortunately, there are ways of dropping this price point for Amateur Radio network interfaces. The first is to strip the TNC of its intelligence, and second is to strip all non-essential functionality from it. This essentially puts the protocol burden on the host computer, and turns the "TNC" into a glorified, albeit slow, network interface unit (NIU). But, even if you could arrive at a $20 9600bps NIU, you're still looking at a price point of $2083.33/Mbps. [e6]_

The biggest influence will come instead from using old modes and pre-existing equipment in novel ways. For example, an NTSC channel sports a 4.2MHz bandwidth channel through which we can send data. That's an awful lot of bandwidth we have at our disposal!

A suitable video card in a computer is essentially a transmit-only serial shift register, and therefore can be used to transmit data, especially when passed through a converter to ensure NTSC compatibility [e7]_. There are ATV sub-bands on the 70cm frequencies. It should be relatively easy to adopt ATV equipment such as PC Electronics' TC70-20Sa ($579.00) [9]_ for application in broadband digital communications. Even with the approximately 20% overhead NTSC framing overheads, it ought to be possible to deliver 3.84Mbps throughput at a minimum (based on the Commodore 64's video framing, clocked at 8.064MHz [e8]_; 320 x 200 monochrome pixels, 60 frames per second). As we can see, at a price-point of only $150.78/Mbps, it's actually cheaper than the DSL price-point by a factor of 2.4 [e9]_.

Provided ATV equipment can work with a flat 6MHz-wide baseband signal (e.g., doesn't depend on the presence of sync pulses for proper operation), it is even possible to achieve up to 12Mbps throughput with careful waveshaping, thus driving operations costs down even further. Employing wavelet modulation promises still higher throughputs while reducing overall equipment complexities.

Quick and Easy End-User Deployment

Another requirement of deploying NgARN is low fuss. Most people just want to pull the thing out of the box, set it on the desk, turn it on, and it just works. Equipment must be designed with this in mind -- particularly driver software. A network interface that essentially emulates a really fast RS-232 port via USB (e.g., via the FT-245 USB host interface chips, for example) perhaps would be the best solution from a price standpoint. Software can talk to it through a trivial interface, and the USB infrastructure provides the necessary DMA operations, thus keeping hardware costs down while minimizing software loads.

The greater the distance between end points, the higher the antenna gain required to meet a minimum signal level, and therefore, the more precision is needed in aiming it. Additionally, due to natural forces, the higher the gain, the more likely it is to fall out of alignment with the other end station. Weather effects of the microwave bands need also be taken into consideration.

Another aspect of low-fuss operation is scalability. 802.11 networks have been shown to scale relatively poorly as the number of nodes increases, and in particular as node distances increases [11]_. Both of these are directly related to the use of MACA-inspired, semi-reliable handshaking that occurs on the channel [e10]_. This argues for a micro-cellular approach towards delivering service. However, this works only in areas with sufficient amateur populations to justify the expense of erecting such microcells.

For these reasons, omni-directional antennas on the lower bands (e.g., 70cm or 2m) is expected to be used to cover significantly larger areas with less power than 802.11-based technologies. Moreover, AX.25 appears to be more link bandwidth efficient than 802.11's layer 2 protocol, especially since it supports proper windowing [e11]_. This seems to suggest that by using a more efficient layer 2 protocol, we are likely to be able to overcome the scalability problems when using 70cm or 2m bands for wide-coverage NgARN LANs.

Advancing the State of the Art

Using the NgARN as an emergency network for commercial Internet systems fulfills only one of the legal requirements for the justification of amateur spectrum. For many, this will be sufficient, as they see NgARN as merely a component in a larger landscape of Amateur Radio activities. For others, however, this is inadequate; as section 97.1(b) states that we should also advance the state of the radio art. The very development of an all-wireless internetwork of course satisfies this requirement to a limited extent, but once established, where do we go from there?

As indicated earlier, the Internet has a number of problems. However, there are potential solutions to these problems, which in my eyes, have gone largely unaddressed. While absolutely none of these are radio-specific, the fact that it is being used *over* radio channels makes them directly relevant to the radio state of the art, particularly in the fields of maximizing bandwidth utilization and encryptionless authentication mechanisms. Moreover, their combined use on a single network constitutes a degree of technological advancement that I haven't seen anywhere else, wired or wireless.

Spam Prevention

Current e-mail systems make it effortless to send messages, while expensive to receive them. Messages consume resources on the recipient's ISP, not on the senders. This is the fundamental basis for spamming [e12]_. But what if we were to reverse this situation completely, and make it expensive to *send* messages and effortless to receive them?

The IM2000 concept [12]_ promises a new way to handling mail, which does exactly as I described earlier. Unfortunately, large-scale deployment has not occurred yet, due to the omnipresence of SMTP and relative lack of mail client software. It appears that the latest development of the IM2000 concept is related more to the Jabber instant messaging system than to the Internet e-mail problems it was originally designed to solve.

Address Space Shortages

It is pretty widely accepted that the Internet is in a state of uneasy balance, what with the use of NAT serving to help mitigate the shortage of IP addresses. However, this likely won't last long. NAT has slowed, but not stopped, the continuing consumption of public IPv4 addresses. According to [13]_, with current IP consumption rates, we could exhaust the IPv4 address space as early as 2016, and as late as 2023. [e13]_

Even if the address space isn't in danger of exhaustion, there are more reasons to adopt a larger address space in an attempt to eliminate NAT. NAT has single-handedly exacerbated the spam problem by rendering the use of reverse-name lookup-based "sender authentication" schemes useless. Alternative spam prevention methods have resulted in ISP customers having increased difficulties in sending legitimate e-mail, *dramatically* increased false-positive rates on spam block listings for innocent ISP users (especially dial-up users employing such acceleration software as SlipStream and derived products), creating many countless hours of phone support overheads that could have been avoided [e14]_. ISPs are also now investing in more sophisticated and vastly more expensive mail server hardware to handle the additional processing loads that technologies such as Bayesian filtering imposes, which can delay mail delivery significantly. NAT also strongly encourages the use of name-based virtual hosting, which has problems of its own, perhaps the biggest of which is corporate branding violations [e15]_.

The exclusive use of IPv6 on the NgARN allows for the development of an all-IPv6 internetwork without having to worry about backward compatibility with IPv4. The entire IPv4-based Internet is a proper subset of the IPv6 address space (::AABB:CCDD, where AA, BB, CC, and DD are the 4 octets of an IPv4 IP address. It may also be written as ::111.222.33.44), thus enabling two-way compatibility with the commercial Internet. Therefore, if you're looking to create a dual-network website, for example, you would assign your server two IP addresses -- the NgARN IPv6 address, and an IPv4-compatible IPv6 address. That will allow effortless routing.

Network address auto-configuration can occur using the stateless automatic configuration of IPv6 addresses documented in [14]_ and [15]_, with the link-local address interpretations suitably modified to use Amateur Radio call signs and SSIDs. A suggested link-layer address format will be described in a forth-coming paper on Amateur Frame Relay (AFR), by the author. In the mean time, a quick and quite incomplete sketch of my ideas is available at [16]_ [e16]_.

I should further point out that we as Amateur Radio operators do not *need* IPv6. The number of amateurs in the United States measures less than a million, and a seemingly insignificant fraction of them are apparently interested in packet radio as it is. As there are fewer than 4000 countries in the world, it follows that the IPv4 address space ought to be more than sufficient for the amateur community. Indeed, we have yet to exhaust the ::44.0.0.0/8 network, and currently shows no known evidence of exhaustion.

The development of an IPv6 network is not so much to satisfy *our* dwindling IP supply, but rather, to serve as a technology development platform that, once refined and debugged, ought to be readily applicable to non-amateur fields, just as AX.25 itself has migrated away from the amateur domain towards commercial and military use. It is in these other domains that the IP shortage is felt most.

Voice over Packet

One thing that the amateur community does seem to be perpetually in fear of is dwindling frequency space. With the majority of Amateur Radio use constituting voice communications modes, it follows that cutting their bandwidth usage is the quickest and easiest approach towards preserving our existing allocations while allowing for greater traffic capacities.

VoP is a relatively new technique to help facilitate re-use of an available channel. It's pretty well known that about 50% of a channel's capacity is wasted by "dead air." Just listen to any 2m repeater QSO. Especially for elderly amateur users, who often speak slowly and take extended breaths while still keyed. While VoP cannot affect their health, it can help better utilize the bandwidth we currently have.

This is precisely why most cell phone services have made the switch to digital methods of re-using available channel bandwidth. By eliminating all redundancy from the voice channel, things like extended breaths, long "ummms," and related artifacts of speech, more bandwidth is made available for other users. It follows that amateur services can also benefit from these techniques as well, especially in more populated areas where repeater usage tends to be heavy.

Ogg Speex [17]_, in particular, promises outstanding voice quality and compression ratios. Combined with a packet-based communications channel, a single 19.2kbps throughput link can handle up to 4 2400bps concurrent QSOs comfortably, and up to 7 if you push the channel hard enough. [e17]_ Each QSO would be assigned a unique number, set via the user's radio. It can probably comfortably hold more, if you're willing to wait briefly while other channel users consume some bandwidth.

The method of operation would be unmodified over current analog radios. When you leave the radio unkeyed, it will play back VoP packets as it receives them. If multiple people are keying and talking concurrently, the streams are played back on a first-come, first-played basis [e18]_. This queuing capability guarantees that there is no audible "doubling" that often occurs with traditional repeaters. Everyone hears everyone else. Assuming they've selected the correct QSO ID of course.

While you are keying, the radio is concurrently transmitting packets that contain your voice content, while receiving other packets in between transmissions (remember it is using approximately 12.5% of the available channel capacity, assuming the parameters established above). The result is that as soon as you release the key, you hear what other people have said while you were transmitting. [e19]_

Expanding repeater coverage via linking repeaters is also better accommodated using VoP technology. Indeed, VoP can make the whole argument about whether EchoLink is ham radio or not moot.

It should also be pointed out that pure packet switching has traditionally had problems with latency issues in dealing with voice-over-packet technologies. This is an area that is still rife with research. However, as amateurs, we have an advantage: *we control the layer 2 protocol.* Indeed, I have proposed a refinement of KA9Q's MACA protocol [10]_ called MA/CAPS [16]_, which promises support for isochronous traffic at the local network level, which promises better quality of service support over a "dumb," packet-switched network. These same techniques can be incorporated into DAMA as well. This is just one of the many areas of research that NgARN can help support.

VoP has another side to it, which should grab the attention of those in RACES or ARES. Let's look at our 70cm IP-over-TV approach discussed above. Assuming we have 4.2Mbps of raw throughput, let's shave 40% off for network header and NTSC sync overheads. That still leaves us with 2.52Mbps of raw data throughput. This is faster than a T1 by a long shot, and with 64kbps dedicated to a single voice channel, it follows that an emergency operations center can provide *phone service* for 39 concurrent calls by aggregating audio from each phone into a continuous stream of packets. Using Ogg Speex compression can provide for substantially more concurrent calls. Imagine having toll-quality audio in an emergency setting, where victims can call out to their loved ones. The use of 70cm should provide for sufficient range so that the other end-point can break out the audio streams into individual phone channels, where phone service has not been disrupted.

Instant Messaging

As discussed at the beginning of this article, a local repeater user was complaining and disgruntled about the preference of commercial Internet instant messengers over voice chatting on Amateur Radio by young children. What if you combined the two? Odds are, the kids will be more interested in ham radio.

Which leads us to another potential benefit of architecting the NgARN: increased amateur service retention and maybe even growth rate. [e20]_

Return On Investment

When someone purchases any kind of product or service, they expect some kind of satisfaction from it. This can be measured in different ways: businesses measure the impact of a purchase on their future profits, while home users measure the impact on their daily lives of a purchase. If the purchaser of a product isn't happy with it, the product is discarded or resold to someone else, often at a reduced price, thus further lowering its value.

It is important that the NgARN must have an excellent "ROI." This can come in several forms:

* Technology developed from the development of NgARN can be adopted by non-amateur industries, just as AX.25 has been adopted by industrial and military organizations the world over. This provides a contribution to society, thus fulfilling the secondary directive of the existence of the Amateur Radio service.

This suggests that the technology developed while working on NgARN must be applicable towards today and tomorrow's problems. Often these problems will go unsolved for significant periods of time, due to lack of commercial interest (the ROI isn't high enough), but which nonetheless are important.

* Those working on the NgARN gain valuable experience that translates into the commercial sector, thus hopefully leading to better jobs, and therefore increased income. This has happened with several open source software developers in the past. Judging by the number of Amateur Radio operators in prominent technical positions around the world, it suggests this occurs for Amateur Radio experimenters as well.

This suggests that experimenters maintain a level of professionalism that is appealing to outside industry. Professionally-written journal articles, passionate yet non-partisan about the technology they're working on, participating in standards approval processes, etc. should all contribute towards the employability of an experimenter.

* Content must be made available on the network. This means people need to put up mail servers, websites, games and entertainment, and other services on the NgARN. Without the average amateur having any content to put up himself, the rationale for having access to the NgARN based solely on the premise of it "being cool" and even "just as cheap" just falls apart. People don't like paying for nothing, however little the cost.

Imagine the pride and joy of a person who just retrofitted his Mazda RX-7 to have a 900HP, 3-rotor engine, only to find out that gasoline doesn't exist, and can't find suitable racing tires anywhere. It's hard to justify making a technologically superior product if it won't be used.

Of course, this means that the early adopters of the NgARN will ultimately be responsible for putting content on the network, in some capacity. A quick and dirty way of providing content is to "gateway" commercial Internet websites onto the NgARN. For example, eham.net or qrz.com can provide special versions of their websites that are "dual-network", allowing a single source of content to exist on both networks. Since the commercial Internet version of the material is the same as the NgARN material, only with the addition of push media to help generate more revenue, it follows that proxy servers can be used on the commercial-side of network access to inject such push media, without affecting the NgARN-visible content. This minimizes the amount of work on the webmaster considerably while concurrently serving two markets.

Some Problems Facing NgARN Today, and Potential Solutions

Standards Management

The amateur community has no equivalent to the ITU or IETF, yet it needs one badly, especially with all the technical innovation occurring with HF digital modes and new layer 2 protocol designs. Standards, which take hold in the amateur community, are loose and informal. This increases the chances of incompatibility between implementations of the same basic concept. I also feel this is largely the reason for lack of technical innovation in related areas such as packet radio; without standards, packet radio has been characteristically hard for people to use, especially compared to the commercial Internet offerings, itself built entirely on a consistent repository of standards.

Take PSK31 for example. Where are the standards covering this mode? How about MT63? I literally had never once heard of Throb until I saw it in a program called gMFSK for Linux. For that matter, what about DAMA-over-AX.25? [e21]_ I won't even discuss the utter frustration that most need to go through to interface their computer or TNC to their radio.

Having an independent Amateur Radio standards body that concentrates solely on *documenting* the various interfaces and modes that we use, ranging from layer 1 standards on up, should be a generic goal that isn't strictly NgARN specific. Yet, it is of utmost importance for NgARN to have a central repository of standards that can be openly searched through and discussed, with support for creating new standards as required. [e22]_

Rewriting and copying of a commercial standard, so that we have a "free copy" of it, should be condoned where possible. This is particularly the case considering that you generally cannot obtain even the Bell 202 standards document from ITU without plunking some serious money down, let alone some other standards from ITU. After all, when sending a packet out on 2m AFSK at 1200bps, do you start with 1200Hz tone or with the 2200Hz tone? Are you even sure it *is* 2200Hz? Some websites I've seen indicates it may be 2400Hz [18]_. The former seems more widely documented, and is therefore most likely being correct. But, the rules of thumb concerning FSK modulation indicates that the latter is more technologically correct. One would be forgiven for thinking that nobody knows for sure, except the commercial vendors of the equipment.

One potential system of standards we can adopt is an ITU-T-like standards naming convention, but with an IETF-style standards track process and philosophy, so that the community doesn't get mired down in too much formalism. For example, the letter A would prefix all Amateur Radio standards. The following letters are used to identify the broad classification of the standard. These need not strictly adhere to ITU-T recommendations, but it's probably a good idea to follow their lead where possible. For example, using Wikipedia's entry for ITU-T as a guide [19]_, here is a hypothetical list of standards we might adopt over time:

* AA.1 would identify the basic standards governing the standards body itself.

* AB.1 could define the standard means of expression for publication in TAPR's PSR, while AB.2 could do the same for QST or QEX.

* AE.1 might define the standard for how Amateur Radio call signs appear to a human being for packet radio applications, including SSIDs. AE.2 might refer to AMPRnet's ::44.0.0.0/8 network IP subnet definitions. AE.3 can define the hierarchical address system adopted ad hoc by packet operators for sending e-mail (e.g., **WB9LOZ@W6PW.#NCA.CA.USA.NOAM** [20]_).

* AG.1 might define a system of data compression

used to shrink packet sizes before transmission. AG.2 might define the VariCode used in PSK31 (or, shall I say, AV.4) communications.

* AH.1 could define the streaming audio format used for voice-over-packet.

* The AJ-series of protocols would most likely be used for amateur television-related standards. One of these could perhaps be the framing requirement for digital transmissions over NTSC and/or PAL ATV channels. However, I question the need for this series of standards, since emissions-standards and layer 1 standards appear to be at home in the AV-series.

* The AK-series of standards are intended for standards pertaining to the prevention of interference. This can come in many forms, from operating procedures for various kinds of nets, to hard technical specifications such as cable or shielding requirements, filter design requirements, etc. However, since many of these requirements are defined already in the FCC and related body legal codes, I question its value, and ponder opening this series up for re-purposing.

* AL.1 can define the long-desired, vendor-independent, fully standardized rig/sound card or rig/TNC interface. AL.2 can define standards on safely installing cables that lead to antennas, etc.

* AO.1 might perhaps define standards on what actually constitutes an S-unit.

* The AQ-series would contain signaling requirements. AQ.1 could provide a home for AX.25's digipeater operational requirements, thus allowing a still-further simplification of the core AX.25 standard. AQ.2 could define the signaling requirements used for voice-over-packet. AQ.3 can define the signaling requirements for DAMA in an Amateur Radio environment. AQ.4 can define APRS's message formats.

* AR.1 can define various CW-related standards, like characters used that are not in the ITU recommended character set. AR.2 could define standard non-net prosigns, while AR.3 could define the standard net prosigns in common use. I don't foresee much use of this series of standards, however; it's such a simple operating mode that its complete operation can be covered in only a few pages of text!

* AV.1-AV.4 might refer to 1200bps AFSK, 9600bps G3RUH, 45.45bps RTTY, and PSK31 emissions. Note that these are *strictly* kept to physical layer considerations. That which we call "PSK31" today would be covered by both AV.4 and AG.2 standards. Just to be sure, AV.5 could define a SSTV emission, just to prove that the AV-series isn't strictly restricted to digital modes.

* The AX-series is probably going to be the widest series of standards documented, and of the greatest interest to packet radio operators. AX.25 retains its designation for historical purposes, but it can be stripped bare of all non-essentials. Packet switching semantics through digipeaters can be defined in an adjunct specification, AX.1, while connection maintenance and operational semantics are defined in AQ.1. AQ.3, as indicated above, defines DAMA semantics and requirements, while AX.2 might define its precise mapping into the AX.25 environment. New layer-2 framing formats, if adopted as standards, can also attain AX-series identifiers. This series isn't constrained to layer-2 specifications, however. Standards governing AQ.4 over AX.25 can appear here, as well as standards governing the mapping of TCP/IP virtual circuits to native AX.25 virtual circuits.

* AY.1 can effectively serve as the equivalent to RFC 822 for Amateur Radio e-mail, defining required headers and mail format. AY.2 can define file transfer protocol standards.

The C, D, F, N, P, S, T, U, and Z ITU-T prefixes don't really apply towards Amateur Radio as such, so these prefixes can be used for Amateur Radio-specific standards that aren't covered elsewhere.

Specific versions of the standards would be identified by a year following the basic name (e.g., AX.25/1984 versus AX.25/1989). If no year specifier is given, it is assumed to refer to the most recent revision of the standard.

To facilitate IETF-style requests for comments, I propose the pseudo-standard ARC, or Amateur Request for Comments. For example, ARC.1 might propose the definition of DAMA into AX.25, before it is given a formal standard identifier, while ARC.2 might be an informational paper that clarifies some aspect of AL.1. Where possible, the IETF rules governing the generation and management of requests for comments should equally apply to ARCs as well.

As with IETF standards, all Amateur Radio standards should be freely available for anyone to learn from, implement, or comment on.

Link Speeds

Say what you will, but 2m is currently the most often used band for packet radio in the amateur service. This is due to a huge array of pre-existing equipment. However, link speeds currently seems limited to 9600bps.

The FCC grants 20kHz channel widths for unspecified digital codes on the 2m band. The sidebands created by AFSK over FM results in roughly 66% of that bandwidth going to waste [e23]_. G3RUH modulation provides 9600bps links by essentially using a variation of BPSK on the link, which makes far better use of the available channel bandwidth. However, I feel that it can be improved upon further.

Having access to wideband amplitude modulation, it is possible to use an SSB emission that consumes 18kHz (leaving two 1kHz "guard bands") of bandwidth, thus providing up to 18kbps of throughput off the bat. Use of wavelet modulation techniques promises the ability to deliver higher data rates, potentially up to 72kbps or more, relatively reliably.

Applying this same idea to 70cm ATV, we can start with a base-line transmission rate of 4Mbps. With (perhaps orthogonal) wavelet modulation approaches, rates as high as 16 to 64Mbps should be possible.

Local Network Access Procedures

Link access procedures and topology for local area networks are very closely related. There are so many ways of implementing a local network that it is often hard to choose which one is right for your needs. In order to help choose what's right for your local network needs, we'll go into the different kinds of access procedures "in general," and later, discuss how they apply to specific RF topologies.

ALOHA

This is where a node doesn't bother listening first before he transmits. If a node has something to say, it'll say it. Whether it actually gets received correctly isn't of any concern to the node per se. Reliability mechanisms are left for higher network layers, such as TCP.

Due to the lack of consideration for other nodes on the network, collisions occur frequently, resulting in lots of lost packets. Link utilization efficiency typically peaks only at 18%. This means, for example, given a 100kbps link, you're only effectively using 18kbps of it. The rest is consumed by collision resolution.

Slotted ALOHA

If nodes can all agree to transmit within certain time slots, then there is an inherent time-based multiplexing that helps avoid collisions. But, how does a node know which slot to transmit in? It just guesses -- it initially picks the next slot in time that is coming up. If a collision occurs, so be it; it is the responsibility of higher network layers to appropriately recover. S-ALOHA has been proven to double link efficiency to 36% peak. S-ALOHA does have the problem of requiring all nodes to be synchronized against a time base, however. This makes sending very small packets (like keypresses in a telnet or SSH application) particularly inefficient, since they take the same amount of time as a very large packet.

CSMA

Carrier Sense Multiple Access is a refinement over pure ALOHA, where the node first listens to see if the channel is clear before transmitting its frame. This is how most humans make use of a simplex frequency [e24]_. If multiple stations accidentally "double" over each other, a collision results, thus causing all packets which have collided to be lost.

Note that radios in the amateur service can typically only transmit or receive, but not both. Hence, collisions are not detected by the transmitters, only by the receivers. This results in some waste on the band, as even a slight collision, affecting as little as a single bit of data, can result in all the time spent transmitting a frame to go to waste. As a result, link utilization efficiency typically peaks at only 36%. The simple act of listening before you transmit doubles the link efficiency over pure ALOHA, and competes favorably with S-ALOHA without the complicated time base requirement. Thus it has better small-packet performance than S-ALOHA.

CSMA/CD

If we were to somehow add in the ability to have a transmitter detect when a collision is occurring, the transmitters involved can abort their frame transmissions early, thus helping to recoup some of the bandwidth that otherwise would be lost. Typically, CSMA/CD networks can theoretically achieve link utilizations as high as 60% under some circumstances, but more often achieve only 36% in real-world use [21]_.

CSMA/CA

In the radio world, it's often very difficult to detect a collision because there is no convenient way to both transmit and receive concurrently. Therefore, alternative approaches are taken to reduce collisions as much as possible. One approach is to coordinate with network neighbors on when it is acceptable to transmit.

CSMA/CA favors collision *avoidance* instead of collision detection. There are two approaches towards achieving this: handshaking and random, pre-emptive back-off delays.

The handshaking approach involves two network nodes employing "request to send" (RTS) and "clear to send" (CTS) packets. The sending node does this by using pure-CSMA to transmit request-to-send (RTS) packets to the destination. RTS packets basically represent an attempt to allocate a chunk of future bandwidth. Once the neighbor receives an RTS packet, a clear-to-send (CTS) packet is transmitted in response. This packet actually is the authoritative packet that grants the network resource to the other node. Upon hearing a CTS packet, the sending station is relatively assured that there will be no collisions while transmitting its data, since in theory, all other listening stations are aware of the bandwidth allocation too.

Because CTS and RTS packets are so small relative to the average data packet, any collisions with them are relatively cheap, and link utilization should be higher than even for CSMA/CD. Theory and reality are only theoretically related, however. It turns out that, while CSMA/CA does achieve slightly higher link utilization than CSMA/CD, in actual practice, you're not likely to notice any significant difference. According to [22]_, the throughputs for handshaking CSMA/CA (what in the paper is documented as RTS/CTS) is about 40% link utilization, which is the theoretical *maximum*. As usual, you can effectively halve that number for real world predicted performance.

Things look brighter for non-handshaking CSMA/CA, however. In this form of collision avoidance, the transmitter checks to see if the link is in use, and if not, *it waits a random period of time before checking again.* If it's still clear, then it will transmit its packet. The odds of two or more nodes transmitting a packet at the same time, and waiting the same random delay, approaches zero. This form of CSMA/CA achieves higher throughput due to not requiring a 3- or 4-way handshake for each packet transmission. Link utilizations can achieve up to 60% [22]_. But, in this case, you still end up *deliberately throttling* your own link; thus, this type of CSMA is beneficial only if you have a lot of data transmit in bulk, and not particularly good for exchanging small-sized packets [23]_.

Token Passing

Imagine there are three people on the local repeater, Alice, Bob, and Cathy. You just tune into the repeater, and hear Bob talking to Alice about something. Then you hear Bob finish up, "Over to you Cathy." Cathy will then say what she needs to Alice, and then to Bob. When Cathy finishes, she finishes up, "Back to you Alice!" Alice then does the same: replying to Bob's comments, and then to Cathy's. "Go ahead Bob," she says as she completes her transmission. This is often called a "round-robin net" or "round-robin rag chew," or even more simply, just a "round-robin," since it's neither a formal voice net nor an ad hoc rag chew.

This is the essence of *token passing*, where each node on the network takes turns using the link. Until a node is possession of the token, anything it has to say is queued up. When it finally does get the token (aka permission to speak), it sends whatever is in its queue [e25]_, following up with another token [e26]_, so that the next node knows it is OK to transmit.

The big win of token passing is that it allows *near 100%* link utilization efficiency. This makes sense when you realize that it is nearly impossible for more than one station to collide with another. Without collisions, there is no need to wait around for the end of the current frame, or wait some amount of random back-off interval. Perhaps most importantly, there is rarely any need to monitor link state at all. A node is always in receive-mode, until it is in possession of the token.

Token passing results in a network that can handle very large and very small packets (and everything in between) equally well, with minimum delay, and minimum complexity.

There are a few link-layer protocol details to work out, however. An arbitration/negotiation procedure needs to be defined in the event that a token gets lost. For example, if Bob is about to transfer control of the frequency to Carol, and the token gets destroyed in a burst of noise, then Carol will never see the token. Bob will, after not hearing anything from Carol in some time, attempt to repeat the token. Hopefully next time, Carol will see it.

In the event that Carol disappears all together (say, after four token retransmission attempts), Bob will need to be aware of who comes after Carol (in this case, Alice), so that the token can be addressed accordingly. This really isn't much of a problem, fortunately. If we overload a local network frequency with 200 nodes, each having 8 octets for layer-2 addresses that total only 1600 octets of a computer's memory.

Now let's suppose that Carol comes back into the network. Since she "missed out" the last time, all the nodes on the network are likely to have dropped her from their node lists. Carol will need to break back into the network somehow. In a wired token-ring network, this is accomplished without interrupting network traffic using physical relays to reconfigure the network [e27]_. However, in a wireless environment, CSMA or ALOHA will be required to inject the "Break" packet needed to grab someone's attention, just as a human operator does to break into a voice round-robin net. If Carol times the break transmission so that it is transmitted between Bob's token to Alice and when Alice starts to transmit, it's likely that at least one of them will hear Carol's break-in attempt. Through this process, other nodes will become aware of Carol's presence, and thus, update their internal ring configuration information accordingly.

If full-on collisions are permitted, break-ins can occur at any time, which will disrupt normal traffic flow just long enough to allow new stations to come on board at any time. The nice thing is, collisions will tend to only occur when stations join the network, which is expected to be a fairly rare occurrence in the grand scheme of things. Thus, while a collision-tolerant token passing network will exhibit somewhat lower performance over a non-colliding token passing implementation, the utilizations are still expected to be well into the 95% or higher region over the coarse of normal operation.

Another detail: what happens if Dave signs onto a frequency, but doesn't hear anyone else in some time? Dave's node might want to attempt to "claim the token," so to speak, in an attempt to self-start the network ring. If no other node objects, Dave will effectively have the token and can transmit at any time. In this way, ad hoc networks can be set up. But if Carol is already listening, and already has the token, and Dave attempts to claim it, Carol can object. At this point, a network is established: both Carol and Dave are aware of each other, and start exchanging tokens. Alice and Bob would join using the normal break-in procedures, since they would be able to hear tokens being passed, and thus know that a LAN has already been established.

DAMA

Let's now consider an alternative "taking turns" approach to networking. Suppose that Alice and Bob can hear each other, and Cathy and Dave can hear each other, but the two groups are deaf towards each other. How would they communicate with each other to form a logical network?

There are two approaches, both of which assume Alice and Bob form one LAN segment, while Cathy and Dave can form another LAN segment, and a router exists between them. If Alice and Bob are using the same frequency as Cathy and Dave, however, the router node is strongly likely to encounter collisions between the two networks.

If a cellular approach can be coordinated between the two groups where Alice/Bob and Cathy/Dave are on different frequencies, then the router can be configured with two radio links, and switch packets between them accordingly. But this is pretty expensive for the router, since now it needs two antennas and two radio connections. It'd be much cheaper to re-use a single frequency, like a *repeater*. Indeed, the wide spatial separation between amateurs with digital equipment is likely to result in "one-man cells," which are pretty useless. [e28]_ Having a centralized repeater is probably most cost effective for providing NgARN service to a wide geographic area.

The repeater itself would need to coordinate with the nodes as to when they are permitted to upstream packets to it. It does this by periodically polling the nodes in the network. For example, again assuming Alice, Bob, and Cathy are using this special repeater, the poll sequence might be to poll Alice for traffic first. If Alice has traffic, she transmits a small handful of packets. After Alice keys down, the repeater then polls Bob. Bob may not have any packets to send this time around, and so transmits a special packet telling the repeater so. The repeater than polls Cathy, etc.

Note that this is essentially implemented with token passing, but instead of a logical ring, a logical *tree* is formed, with the repeater at the root of the tree. The leaf-nodes only pass tokens back to the repeater node. The repeater can then pick and choose which leaf-node it wants to poll next.

Most commercial satellites make use of DAMA in some capacity. Also, DAMA closely reflects how formal voice nets are performed on the air too. For a brilliant example, listen to the HHH net on 7.235MHz. Another good example of DAMA in action is your desktop PC's USB bus.

TDMA

The problem with DAMA is that it has about twice the "control overhead" as token passing, since a token needs to be passed around the network twice [e29]_: once from the tree root to the leaf node, and again from the leaf node back to the root. What if we could arrange for a way to *pre-arrange* when a node can transmit, so that no explicit notifications over the air need take place?

Consider EME QSOs; the Moon isn't an active entity like most satellites are. As a result, with everyone trying to access the same planetoid [e30]_, who or what is to coordinate who has permission to transmit? If a central radio station on the moon did exist, DAMA would be infeasible because the poll cycle alone would take 1.5 seconds *per attendant node*. This might be sufficient for sending e-mail and picture files, but is hardly adequate for even the most basic of telemetry. And forget all together about video conferencing!

The easiest approach to ensuring reliable medium access is to work according to a schedule. Most EME operating procedures I've seen recommend transmitting for 60 seconds, then listening for 60 seconds. This reciprocating behavior is called *Time Division Multiple Access.* [e31]_

Statically allocated time slots, as used in T1s, are quickly falling into disuse due to their inherent inflexibility and relative bandwidth inefficiencies. So this section will discuss the dynamic variant of TDMA, as used by the DTM [e32]_ protocol.

TDMA works by dividing network bandwidth into chunks of time, kind of like slotted ALOHA. Each major chunk of time is called a *frame*, and frame boundaries are isochronous -- that is, you can pretty much tell the time of day by their periodicity. You can emulate this by having a computer send out 1500-octet Ethernet frames like clockwork.

Unlike slotted ALOHA, however, bandwidth allocations for each network user exists *within* each frame, usually divided into slots of much finer granularity. For example, a DTM slot is 64 bits (8 octets) wide. To re-use our hypothetical Ethernet example from the preceding paragraph, 10Mbps Ethernet frames can carry at most 1500 octets. It follows that it can carry at most 187 integral slots. With each frame, and therefore any single slot, occurring approximately 1.2ms apart, each slot therefore represents a single "channel" of approximately 53kbps. Hence, each network user receives a guaranteed minimum of 53kbps of bandwidth, and a potential maximum of 10Mbps, depending on how many slots in a single frame they use. This assumes that frame schedules are monotonic -- each user gets at least one slot in each frame. Thus our Ethernet-based TDMA experiment supports a maximum of 187 users.

As with slotted ALOHA, there is a central control node. Its job is to serve as the network's time reference. All other nodes on the network receive these "syncs" concurrently. Therefore, they are all synchronized against each other.

A node which has data to transmit, then, simply *waits* until its slot(s) roll around in the current frame. When the required number of slots have elapsed, it just transmits whatever data it needs to, for however many slots it has access to. Note that slots need not be adjacent to each other in time!

But, how does a node on the network know what slots it's allocated to use? In slotted ALOHA, you don't. In the case of DTM, each node is statically assigned a "control" slot, which it uses to dynamically request network bandwidth (like a CSMA/CA RTS packet). In the next frame, the controller will either approve or deny the request, depending on the existing network bandwidth consumption status (analogous to a CSMA/CA CTS or NAK packet, respectively). If granted, the information contained therein will include which slots are allowed to be used.

Note that all communications are done through slots. Since all nodes see all slots, it follows that all nodes equally are aware of the slot allocations granted to other nodes. Hence, unlike slotted ALOHA, no collisions can ever occur since all nodes are synchronized with each other in time. Moreover, since communications channels occurs using time as its multiplexing agent, there is no need for explicit layer-2 network headers; channel identification is inherent in *when* data is transmitted or received. Thus, TDMA, like token bus and DAMA, can achieve extremely high bandwidth link utilizations. In fact, TDMA can, under heavy load, get even higher utilizations than either DAMA or token passing precisely because there is no need for layer-2 headers. [e33]_

TDMA does suffer from a few problems however. First, each node on the network still needs a statically assigned control slot. These slots are always allocated, regardless of whether a node has anything to transmit or not. Therefore, the more nodes on the network, the less overall bandwidth there is for data. Therefore, TDMA seems best suited for backbone links, where random proliferation of end nodes is not likely to occur. [e34]_

Second, to transmit data through a non-control slot, a request must be issued to allocate network bandwidth. As with CSMA/CA, this incurs a round-trip packet delay before data transfer can commence. For this reason, bandwidth tends to be allocated for long periods of time [e35]_, thus making it more suited towards connection-oriented operation instead of connectionless operation.

Local Network Topologies

Now that we have seen a few different methods of re-using a channel to support a larger number of users, let's see how topology affects things.

Shared Bus

This is the topology that nearly all Amateur Radio packet networks take today, and is logically equivalent to a whole bunch of network nodes physically attached to a single wire for both input and output purposes. It is the cheapest possible, easiest to deploy kind of network.

Currently, most packet stations are configured to use CSMA on these kinds of networks. However, due to its relatively low efficiency, alternative layer 2 approaches should almost certainly be considered, especially where 1200 or 9600bps links are still prevalent.

Consider: we often complain about the poor performance of a 1200bps packet link. But if CSMA is utilizing only 30% of that link capacity, it's like having only a 400bps connection. If we were using something like token bus or some other method that grants 80% or more utilization of the medium, it would essentially double available bandwidth without one cent of new infrastructure investment. [e36]_

TDMA promises exceptionally high link utilizations. But, it requires precise synchronization and an absolute guarantee that no network node will ever "babble" (transmit for longer than its time slot). Moreover, with the short slot times involved (53.3ms for 1200bps links, assuming a 64 bit slot time), Tx/Rx changeover delays in the local node's radios will dominate the link, regardless of transmitted frame size. Thus, TDMA is not at all well suited for use on shared bus RF links.

For this reason, token passing should *strongly* be considered, especially for ad hoc networks. When used on a shared bus topology, token passing is referred to as "token bus." With the extremely high link utilizations that token bus offers, competing strongly with TDMA, tripling effective throughput on even the slowest links becomes feasible. Indeed, a radio's transmit/receive switching delays become the dominant factor in maximizing performance on the local network. However, since the frame is the basic unit of transmission and not the slot, Tx/Rx changeover delays are amortized into the delay from transmitting the frame as a whole.

It may sometimes be infeasible to erect a token bus network, due to large geographical coverage issues (e.g., hidden transmitter). In this case, routes to other internetworks can be achieved via a digipeater installation, which employs DAMA. The disadvantage to this is that the digipeater cannot transmit and receive at the same time. Therefore, any packets it receives that are destined for another *local node* must be retransmitted on-frequency, thus dropping throughput effectively in half. Packets which are routed *out* of the local network, however, need not be re-transmitted. Therefore, link utilizations may tend to average out at around 60% to 70%, assuming a good mixture of local and non-local traffic exists on the network. This is still a marked improvement over CSMA-based networks.

Folded Bus

When you tune into a 2m or 70cm repeater, you are participating in a folded bus (analog) network connection. Each node on the network has precisely one input and one output port per link, and the two are kept distinct (just as a repeater's input and output frequencies are kept distinct). This approach has many benefits, and is perhaps *the* most appealing network topology to use for ham radio, as it provides the widest choice of link access procedures.

There are several approaches to managing traffic on a folded bus. It is possible to use CSMA/CA, TDMA, DAMA, and Token Bus.

In CSMA/CA, transmitter collisions would be avoided using an exchange of request- and clear-to-send packets. The only significant difference from raw MACA is that carrier sensing is now practical. Because MACA depends on collisions happening at the *receiver*, not the transmitter, to work, its use on a folded bus is not feasible. As far as I am aware, there is no human analog of this approach to reserving bandwidth on the fly.

Another approach for a folded bus that works extremely well is TDMA. In this approach, the repeater node is responsible for serving as a time-base for all other nodes on the network. Each node would be assigned a time-slice with which it would be allowed to transmit on its output. Since reservation grants are broadcast over the bus, all receivers are aware of what reservations are allowed to each node. Note that commercial DTM networks employ folded buses. However, all this assumes that Tx/Rx changeover delays are manageable.

DAMA comes into its own with a folded bus as well, as it is no longer necessary to repeat all local traffic over the same frequency (the analog repeater functionality performs this task automatically). Therefore, DAMA can compete admirably with token bus when used with a folded bus.

Finally, token bus can actually be implemented with a plain-vanilla voice repeater, although having a specialized repeater is still preferred for best performance (e.g., signal regeneration). Token bus over a folded bus topology will completely eliminate the hidden transmitter problem, precisely because all nodes on the network are within earshot of the repeater (and therefore, *logically*, with each other).

Considering that token bus can work with or without repeater assistance, it is probably the best overall choice for a link access procedure for use in NgARN capable networks. It provides maximum compatibility with existing equipment, while providing maximum link utilizations. However, DAMA does have the advantage of supporting prioritized traffic far better than token bus, and is even capable of supporting isochronous traffic.

Star Topology

If each node on the network has a dedicated link (link frequency, physical path, etc.), the result is something like an ATM or 100-base-T network, where all the network nodes connect to a central switch. In this topology, you have maximum aggregate bandwidth, because each link can be utilized independently of the others. Also, link access procedures really don't apply, since multiple nodes do not share the link. [e37]_ Framing can be as simple as a single start bit.

Star topologies for radio networks are expensive, however, due to the need for a complete radio setup per link, including antenna. Moreover, interference is still possible, since a star topology implemented with low microwave or lower frequencies isn't truly point-to-point. Diffraction and overshoot effects can result in a more diffuse coverage area, rather than a straight, explicit line between two points.

One approach of achieving a true star topology is to use *optical* interconnects. Free-space optical networking is a practical reality, as evidenced by the increasing adoption of the Ronja in Europe [25]_. Ronja provides a cost-effective, full-duplex, 10Mbps link that is point to point. Using a single, high-brightness LED, it boasts an impressive 1.4 km (nearly 1 mile) range. With suitable modification to use either additional LEDs or use of a laser, additional ranges are conceivable.

Token Rings

If you distribute the expense of having multiple antennas to each network node, you can arrange each node to have two or four antennas, maximum. Each link would then be connected to an adjacent network node, thus forming a *physical* ring. There are two approaches towards managing traffic on this topology: token ring, and Packet Insertion Multiple Access.

Token ring is, as its name implies, a token passing method, and therefore shares with it all the benefits of token passing. It is a physical embodiment of the logical ring structure. Therefore, each node needs at least two physical links: one to its predecessor node, and one to its successor node. Tokens usually travel in a single direction on a single ring. Self-healing rings require two sets of links, each forming counter-rotating rings.

Because large-scale token ring networks are often big enough so that one packet can finish transmitting before it comes around to the sending node again, the technique of tacking the token onto the end of a frame sequence is used to help "fill up the ring" with useful data. This is called Early Token Release. Since nodes see tokens always at the end of the packet train, they "tack on" their packets at the end, while the remainder of the ring is still processing earlier packets. This drastically improves efficiency. Note that ETR is not possible with token bus.

Packet insertion multiple access (PIMA) is unique to ring-structured networks. In most token ring networks, a node waits for the token before transmitting a packet. Since tokens appear most often at the ends of frames, it follows that nodes always *append* their data to the existing packet train. This can introduce unnecessary jitter in some packet delivery times.

Imagine a continuous train of HDLC frames, with the usual $7E octet being used for inter-frame synchronization. That sync octet can be treated as a token, such that if a node as a packet to transmit, it opens a whole in the packet train right at the next $7E octet it receives, where it inserts (hence, the name, Packet Insertion) its own packet. While transmitting its own packet, it buffers the incoming packet stream. Once its done with its own packet, it resumes the original packet train. The result is the node's packets take priority, at the expense of introducing additional jitter into subsequent packets.

Both PIMA and token ring with early token release are closely related, differing only in where new packets get injected into the packet train.

However, token ring is difficult to deploy because of the requirement that each node be linked to its adjacent nodes. Adding or removing nodes therefore becomes quite a process to be dealt with, making sure that links can be reached, etc. Therefore, token rings are best used for backbones, where node arrivals and departures are infrequent. Moreover, without a self-healing, counter-rotating token ring configuration, it only takes one node failure to bring the entire ring to its knees.

Conclusion

I believe that the deployment of a next-generation Amateur Radio internetwork is not only technologically feasible, but in our best interest. The tools to do so are here now, including for offering multi-megabit class service to large, geographical areas. I've demonstrated how new link access procedures can help utilize our existing resources much more effectively. Finally, I've identified several areas of research, by no means exhaustive, which Amateur Radio practitioners can contribute towards improving the quality of commercial and academic Internet installations. The NgARN is a big project, and will take research and development at all layers of the network stack to achieve acceptable results.

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Endnotes

.. [e1] Ok, ok, I'll admit -- *sometimes* I'll even listen to short-wave radio too. But, don't tell anyone.

.. [e2] You might think that OSPF and other advanced routing protocols will tend to "route around" the failure, as is often promised by the proponents of packet-switched networks. This is true only to a certain extent, and only if there are multiple layers of multiple switches or routers, arranged in a (nearly) fully meshed subnetwork. In other words, an ISP would need to invest an immense amount of money into a self-healing infrastructure, for very little benefit during "normal" operation. Therefore, most do not. After all, how often have you had routing troubles with the @ Home network, or with AOL? Probably pretty rarely. When it does happen, though, it usually takes many hours to repair the broken link.

.. [e3] If we could somehow establish a virtual circuit that is *unreliable*, something mimicking ATM's virtual circuits for example, this would solve the jitter problem while still providing a distinct and persistent path through a NAT barrier. But, then, one must ask themselves, especially in light of MPLS, what is the technological advantage of an IP-based network over an ATM network?

.. [e4] I had a friend review this document, and expressed concern that I was out to redefine the Internet single-handedly -- to make it obsolete, to offer a "competing product." While I'd love to be able to exercise that kind of power, this is unrealistic. As expressed at the opening of this document, NgARN is not designed to replace the commercial Internet, but in part to *influence its evolution* as well as to provide a backup infrastructure in the event of a critical Internet failure. Its separation from the commercial Internet exists only to provide an exercise in self-sufficiency as well as to be free from prior commercial (read "backward compatibility") restrictions.

.. [e5] Actually, that figure is valid only for the first year. I'm guessing here, as I'm unable to find published metrics on account ownership durations, but I expect the average lifespan for any given DSL account is on the order of 3 years. Therefore, the sum total expenditure across those three years is $367.63 + $718.80 (24 months * $29/95/month), or $1086.43.

.. [e6] Indeed, the simplest possible packet setup, involving only 1200bps data links and just the radio, the computer, and the intervening cables (let's guess at $5 total), still doesn't approach the sweet spot price point. $5 / 0.0012Mbps = $4166/Mbps -- *more* expensive than having a cheap, dedicated, 9600bps NIU.

.. [e7] A dedicated piece of equipment, possibly fed via a USB connection via an FT245 chip, will almost certainly be cheaper than an SVGA-to-NTSC scan converter and second video card, however. This is especially true when you consider that some microcontrollers are fast enough to drive monochrome NTSC video at a solid 4MHz to 8MHz entirely in software.

.. [e8] Interestingly, if you use an 8.4MHz dot clock instead of an 8.064MHz dot clock, you can pack 53 octets per horizontal scan line. This just so happens to correspond to the 53-octet size of an ATM cell, for a total data rate of 5.088Mbps, assuming a 53 octet by 200-line display. Taking the cell tax into consideration yields 4.58Mbps of usable layer-3 traffic bandwidth.

.. [e9] If we amortize this out over the same three-year span that we did for Verizon's DSL service above, the difference becomes asymptotically bigger, at only $50.26/Mbps/yr!

.. [e10] I'm forced to wonder if this problem would be as bad had they stuck with KA9Q's *original* description for MACA [10]_, which is a purely unreliable MACA implementation.

.. [e11] 802.11b requires that each packet transmitted to be individually acknowledged, while AX.25 supports up to either 8 or 128 outstanding packets to be sent before waiting for an acknowledgement. Moreover, AX.25's header overhead competes quite favorably with *wired* Ethernet, while 802.11b has significantly more overhead, thus eating into the 11Mbps raw bandwidth.

.. [e12] The word "spam", used in the context of Internet e-mail or Usenet postings, literally is an acronym, standing for *Single Post Across Many*, and originated in the days when Usenet news was most popular. It referred to people who posted a single copy of a message across multiple newsgroups, a (mis)feature of Usenet that was all-too-easily exploited. Spam has largely caused Usenet to virtually become unheard of these days.

.. [e13] To be fair to the naysayers, however, every predicted shortage has so far failed to materialize. Nonetheless, /8-networks are continuing to be sold at a consistent rate. Shortage predictions are always based on *current* consumption rates.

.. [e14] If you've ever found yourself able to dial into your Internet provider, successfully browse the web, but find yourself unable to send e-mail but can receive e-mail just fine, now you know a likely reason why! You can verify this by going first to **http://www.whatismyip.com**, record your current IP address, and then go to **http://www.dnsstuff.com** to do the spam database lookup. If your IP address appears marked in red on at least one blacklist database, this could be the reason why your e-mails are being refused.

.. [e15] For example, doing a name lookup of www.complexdrive.com yields an IP address of 209.126.254.29. Doing a reverse name lookup of that IP yields dish3101.net.ibizdns.com).

.. [e16] Although I champion the use of AFR as a future layer-2 technology for NgARN, it is by no means dependent on it. Everything in this paper can also be achieved with the use of AX.25 for layer-2 transport.

.. [e17] Theoretically it can handle 8, but I'm leaving some room for network overhead fudge factor, since header overhead always eats into the raw bandwidth of a channel.

.. [e18] Note that within a single stream, however, the receiver must be prepared to handle packets that arrive out of order. Unless something that guarantees packet delivery order is used, such as wireless ATM.

.. [e19] Note that we run the risk of a run-away effect where by everyone is transmitting at the same time while still receiving the other participants' packets. Thus, everyone ends up talking about old information, and gets out of sync with each other. Fortunately, there are two ways around this. One is to just wait your turn, as with current approaches. Another is to indicate the channel is in use when detecting packets from someone else on the same QSO ID. Even in the absence of these measures, it is expected that good old-fashioned experience will yield an operating procedure that works well to avoid this problem. After all, the whole point of replaying queued packets is to avoid doubling, not to overload the channel. Common sense enters the picture at this point.

.. [e20] Indeed, the problem of not having enough bandwidth to support current users is precisely the kind of problem we *want* to have, because that is what drives innovation.

.. [e21] It literally took me about *three solid weeks* of Google searching just to find the original German manuscript describing DAMA, with about two more just to find an English language version that wasn't obviously Babelfished. Most of the links were merely abstracts, and references to a DCC paper which required purchase from the ARRL. I now retain a local copy because I do not want to waste that much time looking for what should be commonly available knowledge again. Other 'standards' can usually be pieced together after a few hours of scouring the web for the little bits and pieces contained on websites scattered throughout the Internet. Still, that's a few hours I'd rather spend reading a formal specification, not chasing dead-ends.

.. [e22] Although I lean more towards following the IETF-style standards track process, there is validity to the ITU-style standards track as well. For a good overview of IETF's process and the problems it is currently facing today, see [2]_. I haven't found any detailed references for ISO's or ITU's process, unfortunately.

.. [e23] From the perspective of digital telecommunications, that is. For other applications, like analog voice, they are beneficial because they produce clearer, more fault tolerant communications.

.. [e24] Really, these ought to be called "half duplex" frequencies, since bi-directional traffic flow *is* possible on such a frequency.

.. [e25] Up to a certain reasonable maximum, obviously. Link hogging is heavily frowned upon in token-passing networks.

.. [e26] Just as DAMA over AX.25 has shown, the explicit token pass can be optimized out [24]_. If stations in the logical ring are aware of the ring structure itself, then, e.g., Bob will know it has the medium as soon as Alice keys down. Likewise, Carol will know to transmit when Bob has finished, etc. The mere act of keying down is sufficient to pass the token, provided legitimate data was transmitted.

.. [e27] The Medium Access Unit (MAU), a switch-like device used to ensure ring integrity, is smart enough to wait for the token to pass before reconfiguring the circuit.

.. [e28] This is especially true in California, where the mountainous terrain combined with marine layer and other coastal weather effects results in bizarre propagation paths. Repeaters that can be hit with 50mW in one locality may require 15W several yards away. Worse, one-way propagation paths are not uncommon.

.. [e29] In some cases, this deficiency can be optimized out by re-using existing network protocol overheads in novel ways. See [24]_ for example.

.. [e30] Although Luna is our moon, it is nonetheless big enough to be a planet in its own right. Contrast with Pluto, which to this day, still incites riots among scientists over whether it can be considered a planet. Even the newly discovered 10th planet (2003UB313) is considered a bona fide planet, due to it being larger than Pluto!

.. [e31] In this text, I present TDMA as if it were invented to solve a problem with DAMA-based solutions. I did this to maintain the reader's continuity through the paper. The reality is, TDMA pre-dates DAMA. The famous "T1" standard, invented by AT&T in 1956, used time-division multiplexing to aggregate 24 telephone calls onto a single, plesiosynchronous digital trunk line.

.. [e32] Dynamically-synchronous Transfer Mode.

.. [e33] If all 187 nodes on our hypothetical Ethernet-based TDMA network were actively utilizing their bandwidth, network efficiency approaches 98%!

.. [e34] Looking at the PACTOR series of protocols seems to confirm this. PACTOR is another great example of TDMA between over a channel shared by two nodes. PACTOR appears to be used more often for bulk transfer of e-mail over HF than for keyboard interactive use.

.. [e35] The practice of "fast circuit switching," however, helps alleviate this by *pipelining* the allocation of bandwidth with the actual data transmission itself. It works by first issuing the request, but telling the controller what slots you'll be using for transmission. In the next frame, you send the data, regardless of the controller's response. Since all nodes are aware of bandwidth allocations, no collisions occur, and works great for local network traffic. But the switch may discard the data thus sent if there are no bandwidth resources to switch to an external network with.

.. [e36] The arguments in favor of CSMA/CD and related technologies often hinges around the argument that, "Bandwidth is cheap." The Ethernet folks gave this as the answer to both frame relay as well as to ATM networking, when addressing the issue of traffic shaping for maximizing the use of existing infrastructure investment. The problem is, however, bandwidth *isn't* cheap, especially when used over a shared medium such as radio. Therefore, it is often far more cost effective to utilize existing bandwidth better than it is to make up for the deficiencies of the system by throwing more bandwidth at it.

.. [e37] This is why 100-base-T networks compete so favorably against 155Mbps ATM networks. Any x-base-T network is star topology, just like ATM networks are. Hence, there is no need for CSMA/CD to slow the link access down.

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