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XEDGE 6000

Voice Service Module

Technology and Product Overview

A briefing document prepared by

General DataComm, Inc.

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EXECUTIVE SUMMARY

This product and technology overview details the Xedge 6000 Voice Service Module (VSM), a Line Interface Module for the Xedge 6000 Multiservice Switch family. The VSM is based on the concept that voice is best served by using the ATM Variable-Bit-Rate (VBR) service category rather than the less efficient Constant Bit Rate (CBR). CBR suffers from permanently allocated bandwidth, which is poorly utilized and highly inefficient. The VSM differs from competing VBR Voice-Over-ATM products by providing a standards-based solution and offering interoperability with like equipment. Competitive proprietary solutions are synonymous with single vendor networks.

The VSM incorporates the ATM Adaptation Layer 2 (AAL2) ITU-T Recommendation, which offers a standards-based method for efficiently adapting VBR voice traffic over ATM networks. Although AAL2 offers significant improvements in ATM network efficiency for voice, interoperability with existing ATM products and networks is also needed. As such, the VSM also supports AAL1 and other leading voice networking features. The improvements in efficiency afforded by AAL2 over AAL1 are further detailed in GDC's paper "Adapting Voice for ATM Networks — A Comparison of AAL1 versus AAL2."

The Voice Service Module (VSM) provides:

- ✓ Voice Compression (32K ADPCM & 8K CS-ACELP), Silence Suppression, Echo Cancellation, Fax Relay and Idle Channel Removal
- ✓ CAS Transport
- ✓ Support for Q.SIG SignallingFor use in Applications requiring High Quality, Standards-based, VBR-rt Voice Transport Services

PRODUCT DESCRIPTION

OVERVIEW

The Xedge6000 Voice Service Module is a leading-edge Voice-Over-ATM Interface for the Xedge6000 family of ATM WAN switches and multiservice access concentrators. A slot controller that resides in any Xedge6000 switch, it accepts up to four full ports of T1 or E1 voice (96 or 120 channels) and provides adaptation to ATM cell flows.

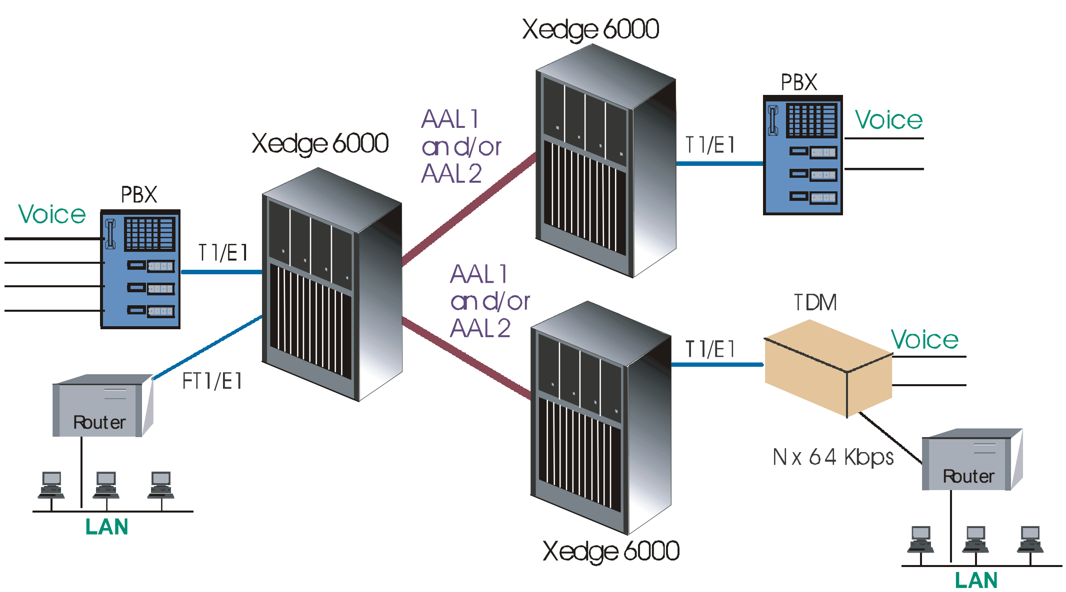

FIGURE 1 – EFFICIENT INTEGRATION OF VBR VOICE

The VSM also provides per-voice-circuit 8K CS-ACELP or 32K ADPCM voice compression, silence detection/suppression, and idle channel removal. It can interpret both Channel Associated and standards-based Common Channel Signaling schemes, such as ITU-T ISUP (Q.761 – Q.764), Q.931, ETSI/ANSI Q.SIG, and DTMF.

Significantly, the VSM supports both AAL1 encoding for backwards compatibility with Structured Circuit Emulation interfaces and AAL2 encoding for standards-based VBR Voice-Over-ATM services.

Many GDC contributions to the standards bodies have been involved in setting the direction for this vital feature. GDC continues to be ahead of its competitors, providing an interface that not only allows backwards compatibility with these devices, but offers future-proof, standards-based, VBR voice internetworking across an ATM network.

KEY BENEFITS

Bandwidth Reduction – AAL2 technologies reduce the bandwidth required for voice traffic. The efficiencies gained via AAL2 lead to extensive reductions in the amount of information transmitted across the backbone, as well as savings in infrastructure costs.

Integration of Voice, Video and Data – For the first time, an ATM product supports standards-based VBR voice, data and video in a single multi-service concentrator with a common ATM backbone.

*Lower Startup Cost – GDC"*s modular approach to Voice-Over-ATM within the VSM allows network designers to start with an entry level CBR AAL1 and VBR AAL2 solution. Value-added voice features such as voice compression and silence suppression can be added as a simple upgrade when required.

Standards-Based – GDC is committed to supporting the latest Voice-Over-ATM standards through a highly programmable firmware environment. The VSM supports both the ITU-T I.363.1 (AAL1) and I.363.2 (AAL2) Recommendations, along with I.366.1 (Segmentation and Reassembly Service-Specific Convergence Sublayer for AAL type 2) and I.366.2 (AAL type 2 Service

Specific Convergence Sublayer for Trunking). I.366.2 extends I.363.2 to provide standard formats for the use of voice compression, silence removal, CAS/DTMF, alarms, and control on the individual channels multiplexed in an AAL2 virtual connection. Therefore, you can be sure that your GDC VSM will adapt to meet new requirements emerging from the standards committees.

Compatibility with Other Vendors' Equipment – GDC's VBR Voice-Over-ATM technology offers far more advanced support for voice applications than most other products on the market. However, the VSM's support of basic Structured Circuit Emulation guarantees interoperability with other vendors' equipment.

High Density – The Xedge6000 VSM offers four times the port density per slot than competing, proprietary VBR Voice-Over-ATM solutions. This means lower costs per port for provisioning voice services over the ATM backbone.

Future-Proof Solution – The VSM is designed to be as future-proof as possible, with pre- dominantly software upgradable ATM adaptation layers and voice processing and signaling capabilities.

KEY FEATURES

√ Flexible physical interfaces:
  –Quad port E1 or T1 digital connections
  –Modular capacity for up to 120 Voice- Over-ATM channels
  –Up to 120 bundles per module
√ Bandwidth efficiency
  –Simultaneous AAL1 & AAL2 support
  –Voice compression
  –Silence detection and suppression
  –Idle channel removal
√ Full range of ATM circuit options: PVCs, SPVCs, SVCs
√ Simultaneous voice and data support

√ Integrated echo cancellation system
√ Automatic FAX/modem bypass or Fax Relay
√ Channel Associated Signaling (AB signaling) plus three narrowband-ISDN signaling options: ISUP, Q.931, and Q.SIG
√ ATM signaling support
√ Trunk conditioning — T1/E1 alarm reporting and monitoring of ATM VC failures
√ SNMP network management

Table 1 – Comparison of Five Voice Service Module Versions

		Description
VSM	T1 and	Basic VSM supporting two or four E1/T1 interfaces, 96 T1 or 120 E1 Voice channels, AAL1 Structured Circuit Emulation, AAL2, idle channel removal, and CAS/CCS signaling support
VSM/48	T1	Basic Module plus 48 channels of Voice Compression/Echo Cancellation and Silence Suppression
VSM/60	E1	Basic Module plus 60 channels of Voice Compression/Echo Cancellation and Silence Suppression
VSM/96	T1	Basic Module plus 96 channels of Voice Compression/Echo Cancellation and Silence Suppression
VSM/120	E1	Basic Module plus 120 channels of Voice Compression/Echo Cancellation and Silence Suppression

PHYSICAL INTERFACES

Quad Port E1/T1 Digital Connections

Each Voice Service Module supports up to four T1 or E1 interfaces, providing connectivity for up to 96 T1 or 120 E1 voice and data channels into the ATM network. A Xedge6640 switch easily supports a configuration of two E3 ATM interfaces, and sixty E1 interfaces, yielding 1800 voice channels in a single, cost-effective, Voice-Over-ATM system.

Up To 120 Voice Channels per Controller

Even though the Voice Service Module is a powerful solution, we realize that there may be network planners with smaller, but just as important, voice and data networking requirements. The VSM has been specially designed to provide maximum architectural flexibility to meet the varying demands of ATM and voice networks throughout the world.

Table 1 shows that, for very simple AAL1-only requirements, the basic VSM provides a straightforward platform for up to four ports of Structured Circuit Emulation over an ATM network, with the added capability to perform both Common Channel and Channel Associated Signaling interpretation. The basic VSM also supports AAL2 capability, including idle channel removal, which provides significant advantages over AAL1 in some cases.

The addition of a single daughter card provides either 48 T1 or 60 E1 channels of VBR voice capability. These voice channels can be selected arbitrarily from any of the 96 or 120 channels available on the Voice Service Module. Even with the daughter card installed, it is still possible to configure all channels as Structured Circuit Emulation and bypass the additional voice processing capabilities, such as compression. Adding a second daughter card provides an extra 48 or 60 channels of VBR voice capability, bringing the total number of VBR voice processing channels to 96 or 120 per card. In this way, network planners can start off with a low cost, entry-level system and be assured of a simple migration path to full featured VBR Voice-Over-ATM for each of the four T1 or E1 interfaces.

Up to 120 Bundles per Module

Voice or data channels with the same destination can be grouped together in a bundle, a single ATM Virtual Channel Connection (VCC) that serves multiple narrowband trunk circuits. These narrowband channels are carried within ATM cells over the virtual connection, and the corresponding narrowband signaling messages may be carried within the same (or a different) bundle.

Bundles provide:

- A reduction in the number of Permanent Virtual Circuits (PVCs) in the ATM network by aggregating multiple voice channels within a single ATM connection
- Bandwidth efficiency through better use of the ATM cell payload
- Reduced delay by increasing the packing density of the ATM cell with multiple channels and therefore reducing the fill time

Each VSM supports up to 120 bundles, with a minimum bundle size of 1 channel, and a maximum of 120 channels. Both contiguous and non-contiguous channels within the T1 or E1 frame can be aggregated into a single bundle (Figure 2).

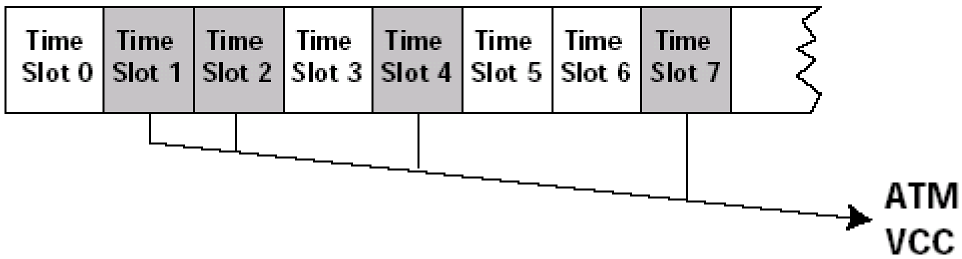

FIGURE 2 – CONTIGUOUS AND NON-CONTIGUOUS TIMESLOTS IN A SINGLE BUNDLE

Each bundle can be configured to carry the number of voice channels appropriate for the particular route being served. With AAL2 support, the actual bandwidth used by a bundle at any time depends on voice activity. For example, voice channels can be dynamically added to or removed from the AAL2 bundle, depending upon whether a phone is on-hook or off-hook. This flexibility to alter the size of the bundle instead of dimensioning by peak rate requirements, enables the released broadband capacity to be used for other (non-narrowband) services.

In fact, the Xedge6000 VSM can take timeslots from any of its four physical interfaces and map them into a single ATM VCC containing up to 120 E1 or 96 T1 channels (Figure 3). The ability to place large numbers of compressed, silence-removed voice channels into a single ATM VCC, to pipe these statistically across an ATM backbone, and to re-assemble them into high quality voice channels at the receiving end is an extremely efficient method of hauling voice traffic over a long distance using a switched ATM network.

BANDWIDTH EFFICIENCY

Simultaneous AAL1 and AAL2 Support

A key feature of the VSM is its ability to support both ATM AAL1 and AAL2 protocols in the ATM network. This ensures that both data and voice services are adequately supported within the VSM and that additional voice features such as compression, silence suppression and idle bandwidth detection are efficiently treated in
the adaptation process.

In fact, not only AAL1 and AAL2 but also AAL5 protocols are an inherent part of the VSM structure. AAL5 carries signaling messages efficiently between voice networking elements and supports call-by-call N-ISDN to B-ISDN interworking in VSM Release 2. Overall, the VSM is a future-safe solution to the networking needs of today that will also address emerging technologies and functions.

Voice Algorithms/Compression

The voice algorithms supported by the VSM include64K PCM (G.711), 32K ADPCM (G.726), and 8K CSACELP(G.729A). This means the user can select quality of service and minimum bandwidth on a per-channel basis. Optioning all channels for minimum bandwidth (8K CS-ACELP), with trunks designed for 80% offhook at busy hour and 50% silence content in active voice calls, yields an effective compression ratio of approximately 12:1. Each incoming voice channel can be individually configured for “compression” or “no compression.”

The future-proofed VSM architecture allows for additional voice compression algorithms to be added or substituted via software upgrade.

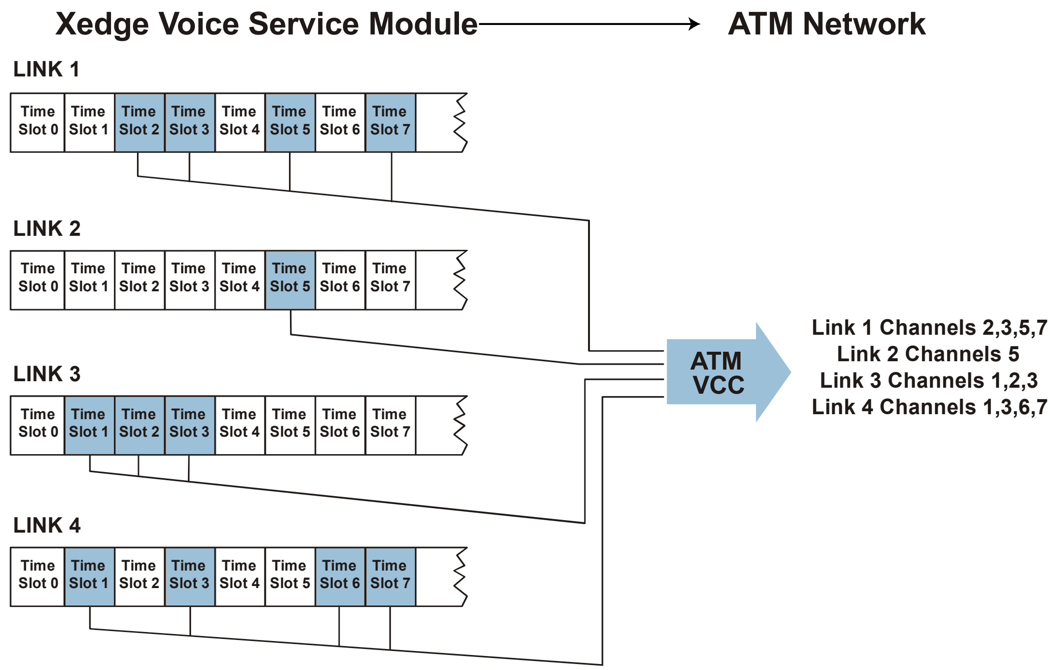

FIGURE 3 – MULTIPLE CHANNELS INTO A SINGLE ATM VCC

Silence Detection and Suppression

In a typical voice conversation, one person speaks while the other listens. There is no need to transmit bandwidth within a bundle when a person is not speaking. An Advanced Silence Detection Algorithm (ASDA) is employed within the VSM to detect and remove silence. When silence is detected, no cells are transmitted. ASDA is disabled when fax/modem tones are detected, when a data call is indicated by ISDN signaling, or when statically configured as such by the network operator.

When a call is placed from a noisy environment, silence detection and removal may be quite objectionable to the listener. The simplest solution disables silence removal unless the background noise level is below a configurable noise threshold. A more sophisticated solution, employed by the VSM, transmits a sample of the background noise at the end of each talk spurt. The background noise sample is then played back repeatedly until voice activity resumes.

Idle Channel Removal

The bandwidth required for voice in the ATM network can be reduced by not transmitting cells when the voice circuits are idle. This provides significant bandwidth reduction over AAL1 circuit emulation.

The VSM monitors the signaling state of each circuit and, when there is no voice call active on a certain channel, signals the AAL2 layer to stop transmitting cells for that channel in the AAL2 PDU. When this bandwidth is freed, it can be used by other applications. A typical example is an enterprise network where voice traffic is very heavy during the day, while at night, it may drop to near zero. A remote archiving program begins to run across the network at night. The bandwidth that was being used for voice during the day is now used for data overnight.

Packet Fill Delay

Xedge6000 VSM uses a 5 ms Packet Fill Delay (Packet Time in I.366.2) for PCM and ADPCM, and a 10 ms Packet Fill Delay for CS-ACELP as specified in I.366.2. These values are used to comply with I.366.2 and offer a standard predefined profile. The VSM conforms with the predefined profile identifier 7 of I.366.2.

CIRCUIT OPTIONS

ATM Circuit Options

The VSM supports the use of ATM PVCs, Soft PVCs (SPVCs), and Switched Virtual Circuits (SVCs) within the ATM network. Basic voice services may use PVC/SPVCs, while more efficient voice networking may use the Common Channel Signaling protocol to dynamically construct end-to-end switched voice calls on demand using ATM SVCs.

Simultaneous Voice and Data Transport

The VSM supports an N x 64 Kbps circuit type for the transmission of data via bundles. On circuits of this type, voice compression, silence detection, idle detection, and fax/modem tone detection are all disabled. N x 64 Kbps data streams can be transmitted using AAL1 adaptation and are end-to-end compatible with ATM devices employing Structured Circuit Emulation for data connectivity.

INTEGRAL ECHO CANCELLATION

In an ATM network, factors contributing to the delays experienced by voice services include:

- Compression Time, the time taken to compress a voice channel down from 64 Kbps PCM
- Cell Assembly Time, the time taken to pack the voice or compressed voice into an ATM cell
- Network Delay, the time taken to transmit the cell from one end of the ATM network to the other
- Decompression Time, the time taken to decompress the voice channel back to 64 Kbps PCM

In voice networks, delay can effect voice transmission quality because of the echo-like effect that results from the reflected signal at the receiving end. While a little delay is tolerable in a normal voice call, delays above a certain level tend to cause an echo that is confusing to the listener and reduces the conversation to a series of mis-timed, half spoken sentences.

In particular, if channel bundles are not employed, the cell assembly delay can be significant even at 64 Kbps. This delay gets much worse when reduced bandwidth (compressed) voice is assembled into cells.
To counteract this phenomenon, voice networks often use echo cancellers external to the network to negate the echo. However, these external units are typically bulky, expensive, or both. An integrated echo cancellation system in the VSM provides the capability to cancel inherent echoes for up to 120 channels of voice simultaneously. The VSM echo cancellation system complies with the ITU-T G.165 and G.168 Recommendations and can be manually configured to “on” or “off” per channel. The VSM echo cancellation function is adaptive, dynamically calculated for each individual channel within the module, and supports a tail-length of up to 20 ms.

AUTOMATIC FAX/MODEM BYPASS

Fax and high-speed modem signals cannot tolerate the signal degradation associated with voice compression. The VSM can be configured (on a per-channel basis) to automatically detect a fax or modem call and to switch the circuit from 32K or 8K compressed voice to an uncompressed 64 Kbps path (with echo cancellation and silence detection/removal disabled). When using 8K CSACELP, a switch to 32K can be selected in lieu of switching to 64K. This improves the fax/modem throughput, but reduces the bandwidth increase.

Fax and modem channels are detected using tone detectors to monitor active voice circuits for fax and modem handshake sequences. When such tones are detected, voice compression is disabled and the circuit is switched to 64 Kbps or the compression is disabled, and the circuit is upspeeded to 32K or 64K. Fax Relay is another method of supporting Fax Traffic on VSM. A carry-over from TDM technologies, Fax Relay (also called Fax Demodulation/Remodulation) extracts user data from the PCM traffic carrying the data, transmitting it across the Xedge network in native digital form (generally at rates 14.4K and below). At the receive end of the connection, the user data is “Remodulated” into PCM form, for eventual decoding at the receive Fax site.

Either technique (Fax/Modem Bypass or Fax Relay) may be enabled on VSM to suit Xedge user needs. The circuit bandwidth remains at 64 Kbps (or 32K with 8K CS-ACELP) until the call is completed. The next call on this circuit returns to the configured compression rate.

Unlike Fax Relay, which adequately compensates for fax traffic but in essence ignores modem traffic, this automatic fax/modem bypass feature maximizes the throughput of both fax and modem traffic.

SIGNALING OPTIONS

Channel Associated Signaling (A/B Signaling)

Channel Associated Signaling (on-hook/off-hook detection) can be interpreted by the VSM and employed to add, or drop, individual channels dynamically from a bundle. The E1 VSM uses Timeslot 16 for identifying and processing the signaling bits. The T1 VSM employs robbed bit signaling for identifying and processing the ABCD signaling bits.

If the bundle is transported over the ATM network using either a PVC or an SPVC, CAS may be used for reducing or increasing the bandwidth of an AAL2 connection by adding or removing an active or idle voice channel from the ATM trunk. When SVCs are used in the ATM network, the VSM can dynamically create ATM connections on a call-by-call basis. Signaling can be configured within the VSM on a per channel basis, enabling support for a wide range of channel types, as shown in Table 2.

Table 2 — Supported Channel Types

E1:	SSR2 Line Signaling Digital Version Q.421 Q.424
T1:	TR 43801 Section C: Sleeve Ground Dial Pulse (SGPO), Dial Pulse (DPO/DPT), E & M, Foreign Exchange (FXO/FXS), Off Premise Extension (OPX),Multi Frequency Signaling, Revertive Pulse (RPO/RPT), Sleeve Dial Pulse (SDPO), Duplex (DX), Private Line Auto Ring Down (PLAR), PulseLink Repeater (PLR), Ring Down (RD)

Common Channel Signaling (N-ISDN Signaling)

Three narrowband-ISDN Common Channel Signaling protocols are supported within the VSM:

- ISUP: Integrated Services User Part, the call control part of the SS7 protocol defined by ITU-T recommendations Q.761 and Q.764.
- Q.931: A signaling protocol associated with public ISDN networking
- Q.SIG: A newer protocol designed for private PBX interconnection

Each T1/E1 interface can be configured for either ISUP, Q.931, or Q.SIG. As with Channel Associated Signaling, bandwidth reduction benefits are derived from processing Common Channel Signaling messages. However, the maximum benefit is derived when these narrowband signaling protocols interwork with ATM SVCs.

Consider a private PBX application where the VSM processes Q.SIG commands and routes individual voice calls over SVCs. In this case, the degree of network complexity is much less than in a more traditional Q.SIG deployment using fixed-bandwidth leased lines. The VSM's ability to place on-demand, switched calls between PBX systems eliminates tandem voice switching and greatly reduces the number of voice trunks in the network.

Support for these narrowband ISDN protocols allows for extended features that are not supported in CAS, such as calling line identification (CLI, or caller ID), clear channel 64K, and switched N x 64 Kbps multi-rate ISDN circuits. The VSM allows these supplementary services to be passed transparently through the ATM network between PBXs.

Through the Q.931 interface, the Xedge 6000 VSM offers a standards-based Primary Rate ISDN connection, and can connect and accept messages from ISDN- compatible devices other than PBXs. For example, a dial-up ISDN router can place switched ISDN calls across the Xedge6000 network as if it were connected to the public ISDN.

ATM Signaling Support

While support for CAS and CCS offers a narrowband signaling interface, it is also important to understand the broadband ATM signaling capabilities of the ATM device. The key to determining if any given ATM product satisfies both current network implementations and future ATM network extensions, is not merely checking that narrowband signaling protocols are supported.

Network planners must determine that the ATM product offers expansion for both narrowband and ATM (or broadband) signaling and assures proper interworking. Networks using narrowband signaling, whether it be

CAS or CCS, are widely deployed. Both product features and standards for these networks are mature and well understood. However, product features and standards for broadband signaling are, by comparison, still rather immature and continuously evolving.

Today, there are four solutions for ATM broadband interswitch signaling:

- Proprietary SVCs
- The ATM Forum's Interim Interswitch Signaling Protocol (IISP)
- The Private Network to Network Interface (PNNI), also from the ATM Forum
- The Broadband-ISDN User Part (B-ISUP or B-ICI in ATM Forum specifications)

The Xedge 6000 product line currently offers support for the IISP and will offer PNNI and B-ISUP based interfaces in the future.

In addition to the existing ATM broadband signaling standards, work within the ITU-T has begun on defining an AAL2 signaling protocol. This AAL2 signaling protocol work will define standards that allow ATM products to not only switch SVCs at the ATM layer, but also provide switched circuits at the individual channel level (CID) within an AAL2 SVC. VSM Release 2 software will support both ATM layer SVCs and AAL2 signaling for the connection of both ATM layer SVCs and individual AAL2 channels.

TRUNK CONDITIONING

The VSM fully supports all aggregate T1/E1 failures, such as local and remote alarms in T1. Failure conditions associated with the full T1/E1 trunks are reported on both ends of the connection and appropriate actions are taken on all channels associated with the failure. VC failures are monitored on the ATM interface and appropriate actions are taken on each individual voice channel in the VC.

NETWORK MANAGEMENT

The VSM is fully managed via an SNMP MIB agent within the Xedge 6000 system. The GDC ProSphere Network Management System fully supports configuration and monitoring of the VSM with an easy to use graphical interface. Additionally, a craft port allows the user to view and instruct the MIB from a local or remote RS-232 interface.

FEATURE MATRIX

Table 3 identifies the capabilities of each of the five Xedge6000 VSM products.

TABLE 3

Features	VSM	VSM/48	VSM/96	VSM/60	VSM/120
AAL1 Support	Yes	Yes	Yes	Yes	Yes
AAL2 Support	Yes	Yes	Yes	Yes	Yes
Maximum Voice/ Data Channels					
2 Ports T1	48	48	48	48	48
4 Ports T1	96	96	96	96	96
2 Ports E1	60	60	60	60	60
4 Ports E1	120	120	120	120	120
Maximum Channels with Voice Processing (compression)					
2 Ports T1	0	48	--	--	--
4 Ports T1	0	48	96	--	--
2 Ports E1	0	--	--	60	--
4 Ports E1	0	--	--	60	120
N x 64 Kbps Voice (AAL1/AAL2)	Yes	Yes	Yes	Yes	Yes
N x 64 Kbps Data (AAL1/AAL2)	Yes	Yes	Yes	Yes	Yes
Idle Channel Removal	Yes	Yes	Yes	Yes	Yes
ADPCM 32K/CS-ACELP 8K VC	No	Yes	Yes	Yes	Yes
Echo Cancellation	No	Yes	Yes	Yes	Yes
Silence Detection/Suppression	No	Yes	Yes	Yes	Yes
Channel Associated Signaling(CAS)	Yes	Yes	Yes	Yes	Yes
Common Channel Signaling (Q.SIG)	Yes	Yes	Yes	Yes	Yes
Common Channel Signaling (Q.931)	Yes	Yes	Yes	Yes	Yes
FAX/Modem Bypass	No	Yes	Yes	Yes	Yes
Compatible w/ATM Forum T1/E1	Yes	Yes	Yes	Yes	Yes
Circuit Emulation Service	Yes	Yes	Yes	Yes	Yes
Downloadable signaling, compression, and adaptation processing	Yes	Yes	Yes	Yes	Yes
SNMP Managed	Yes	Yes	Yes	Yes	Yes

GLOSSARY

AAL1	ATM Adaptation Layer 1 (defined in ITU-T I.363.1), the type of ATM adaptation principally used for circuit emulation services over an ATM network
AAL2	ATM Adaptation Layer 2 (defined in ITU-T I.363.2), a new type of ATM adaptation used for Variable-Bit-Rate Voice-Over-ATM services
AAL5	ATM Adaptation Layer 5 (defined in ITU-T I.363.5), the type of ATM adaptation principally used for frame and packet transport over an ATM network
ADPCM	Adaptive Differential Pulse Code Modulation, a compression algorithm for voice as defined in ITU-T G.726
ATM	Asynchronous Transfer Mode, the cell relay service that transports mixed traffic types over a common communications medium
CS-ACELP	Conjugate Structure-Algebraic Code Excited Linear Predictive, an 8 Kbps Voice Compression algorithm as defined in ITU-T G.729A
DTMF	Dual Tone Multi-Frequency, the signaling system used for push button or Touchtone dialing
ISUP	Integrated Services Digital Network User Part, the call control part of the SS7 protocol defined by ITU-T recommendations Q.761 and Q.764
PCM	Pulse Code Modulation, the basic modulation scheme for transporting voice channels in 64 Kbps timeslots
PVC	Permanent Virtual Circuit, a permanently configured ATM virtual connection
SPVC	Soft Permanent Virtual Circuit, a SVC connection originating and terminating with the Xedge6000 network that appears as a PVC to the endstations
SVC	Switched Virtual Circuit, a dynamically signaled logical end-to-end connection between two end-points in a network, established as needed by the end-user device
VBR	Variable-Bit-Rate, an ATM Traffic Class offering statistical multiplexing gains for bursty applications in the network
VC	Virtual Channel
VCC	Virtual Channel Connection, an ATM logical connection used to connect endsystem devices to each other in a connection oriented network that may be a PVC, SVC, or even an SPVC
VSM	Voice Service Module, General DataComm's new, leading-edge VBR Voice-Over-ATM module for the Xedge 6000 family VTOA Voice Telephony Over ATM

ABOUT GENERAL DATACOMM, INC.

General DataComm, Inc., is a worldwide leader and pioneer in the design, manufacture, and integration of multiservice switching solutions for wide area networks.

General DataComm's Xedge 6000 family of switches delivers cost-effective solutions for public network providers and large private network operators in government, transportation, utilities, energy, and education sectors.