

Product Bulletin

TNETV2520 VoIP Enterprise Gateway Solution

The TNETV2520 solution from Texas Instruments (TI) is targeted at business VoIP applications requiring channel densities ranging from single T1/E1s up to multiple T1/E1s. This solution can be implemented in standalone VoIP gateways or as a VoIP gateway card in legacy PBX or IP-PBX applications. It provides seamless voice and fax communications between traditional and packet telephony systems, enabling business customers to migrate to VoIP.

The TNETV2520 is a full-featured silicon and software solution based on a dual-core 300 MHz TMS320C55x™ DSP coupled with field-proven Telogy Software™ for VoIP to provide voice/fax processing support for up to 64

channels of PCM, 32 channels of low-bit rate voice and two instances of conference bridging with up to 16 user each.

TNETV2520 Silicon Features

The TNETV2520 contains two high-performance, low-power, fixed-point TMS320VC55x™ DSP subsystems. Each subsystem operates at 300MHz and contains dual MACs, two ALUs, three internal read buses and two internal write buses. Both subsystems contain 384 K-Bytes of RAM and a 24K-Byte Instruction cache. In addition, the DSP subsystems share 512 K-Bytes of RAM and 512 K-Bytes of ROM.

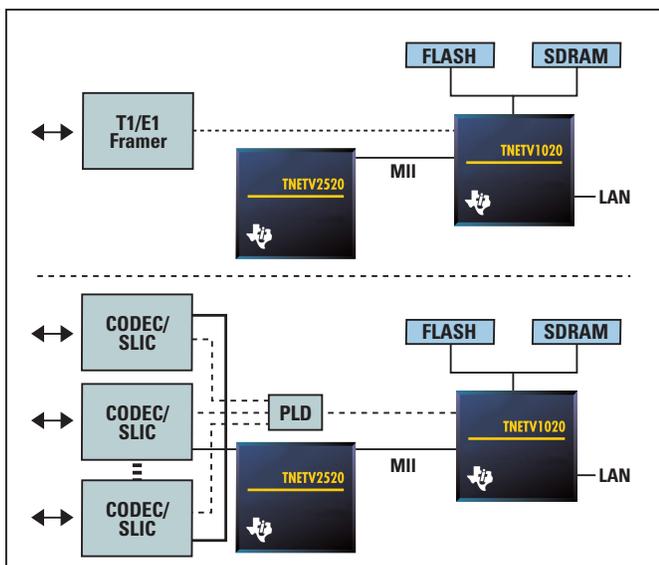
The TNETV2520 has a variety of peripherals to allow system flexibility and customization:

Key Features

- Industry-leading, full-featured silicon and software platform providing a highly flexible and scalable solution
- Field-proven Telogy Software™ for VoIP with over 150 million ports shipped
- Carrier-certified echo cancellation
- Up to 32 channels of low bit rate voice and up to 64 channels of PCM
- Two instances of conference bridging with up to 16 users each
- GMII interface (10/100/1000) and IP encapsulation on the DSP allows flexible scaling to larger channel densities

- Four 256-Channel Multichannel Buffered Serial Ports (McBSP)
- 32-bit parallel Host Port Interface (HPI)
- Universal Test and Operations Physical Interface for ATM (UTOPIA) Level II interface
- 16-bit width up to 150 MHz External Memory Interface (EMIF)
- GbitMAC with GMII Port to allow IP encapsulation on the DSP, lowering the host processor load
- Two 6-pin VLYNQ™ ports
- Two VDMA's to optimize packet voice over VLYNQ serial bus

Single T1/E1



Multiple T1/E1

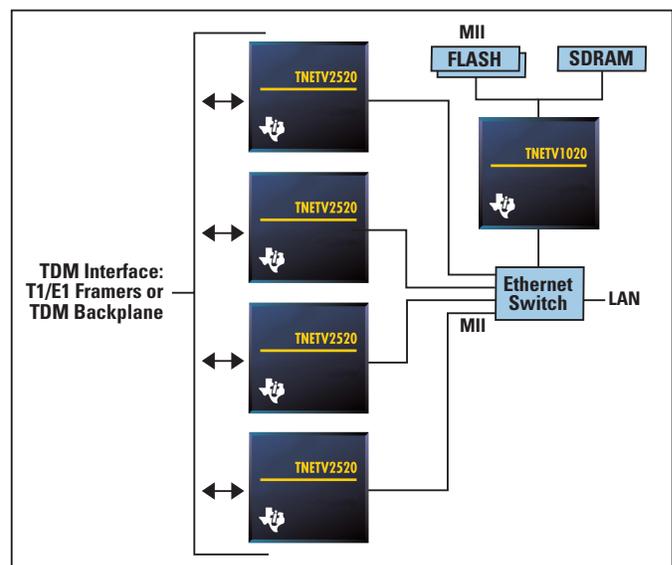


Figure 1

TNETV2520 Processor Architecture

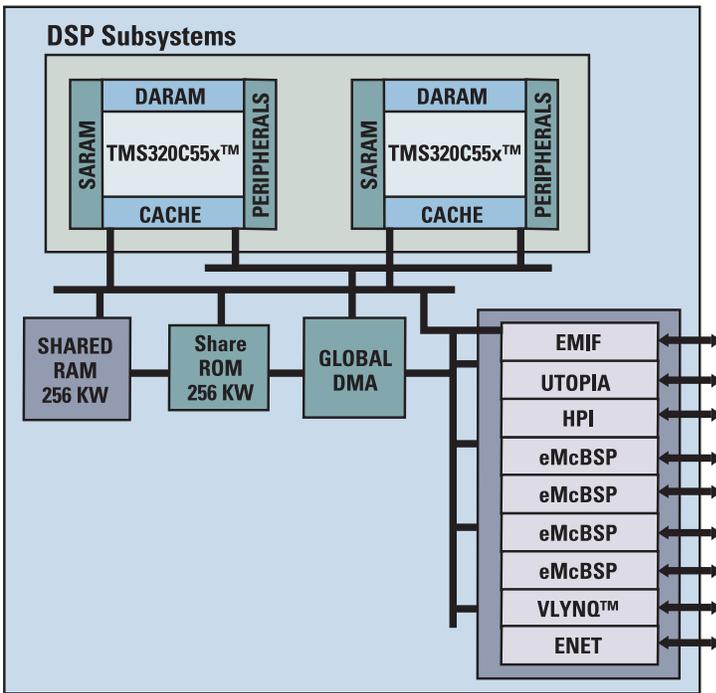


Figure 2

More detailed silicon information is described in the TNETV2520 data sheet that is available upon request.

TNETV2520 Software Features

As one of the earliest pioneers in VoIP, TI began developing products for the enterprise market in the (mid/early) 1990s, when companies started to integrate different types of data on their networks and the proliferation of the Internet presented an opportunity for a universal transport mechanism. With the largest installed base of field-hardened gateway specific solutions, TI offers worldwide communications equipment manufacturers and designers the broadest range of robust and complete voice, fax and video over IP solutions built around TI's award-winning Telogy Software for VoIP.

Key Telogy Software capabilities include:

- Full range of voice vocoders: G.711 (PCM), G.723.1A, G.726(ADPCM), G.729AB, etc
- T.38 Fax Relay: T.30, V.17, V.29, V.27ter, V.21
- V.34 Fax Support

- Voice activity detection (VAD) and comfort noise generation (CNG)
- Line echo cancellation: G.168-2002
- Packet playout, adaptive jitter buffer and lost packet compensation/recovery
- UDP and RTP packet protocols
- In-band tone detection and generation: DTMF, MF, Call Progress, etc
- Caller ID: Bellcore, ETSI, NTT, etc
- Modem over IP: V.152
- Security: SRTP
- Conference Bridging
- Voice quality monitoring, serviceability, manageability and network management

Figure 3 describes the software architecture for gateway solutions. Each box represents a software component required to implement the features for voice, fax, modem, signaling, and network management functions.

Multiple instances of each software component can exist to facilitate support of concurrent, multi-channel operation. Each instance shares common program memory and has unique channel-

specific data memory to maintain information regarding the state of the channel, including network management and diagnostic information.

Conference Bridging

The goal of conferencing systems is to support the greatest number of users while minimizing the voice delays associated with conferencing. Conferencing systems must be able to support any combination of TDM and packet users while not limiting packet users to the same codec. Speaker selection is required for conference bridges with more than 6 participants to provide the expected voice quality in a variety of background noise conditions but the algorithm must be robust enough to guarantee conversational quality of the user's experience.

Telogy Softwares' conferencing solution accepts inputs from all conference participants. It also performs the following functions; performs noise reduction on input signals increasing overall quality of conference, performs speaker selection, and produces composite signals for conference users. The TNETV2520 supports conferences of 16 participants per core (total of 32 participants).

Line Echo Cancellation: Telinnovation™ Solution

The physical media used in today's telephony systems requires a conversion from a two-wire to a four-wire operation. Each conversion calls for a hybrid. These hybrids do not perfectly match impedances, resulting in echo. The goal of the echo canceller is to detect and cancel this echo as quickly as possible. The inherent challenges require that the echo canceller not only distinguish the location of the echo but also be able to distinguish the echo from double talk and other events.

The length of time over which echo is expected to arrive is referred to as the echo tail length.

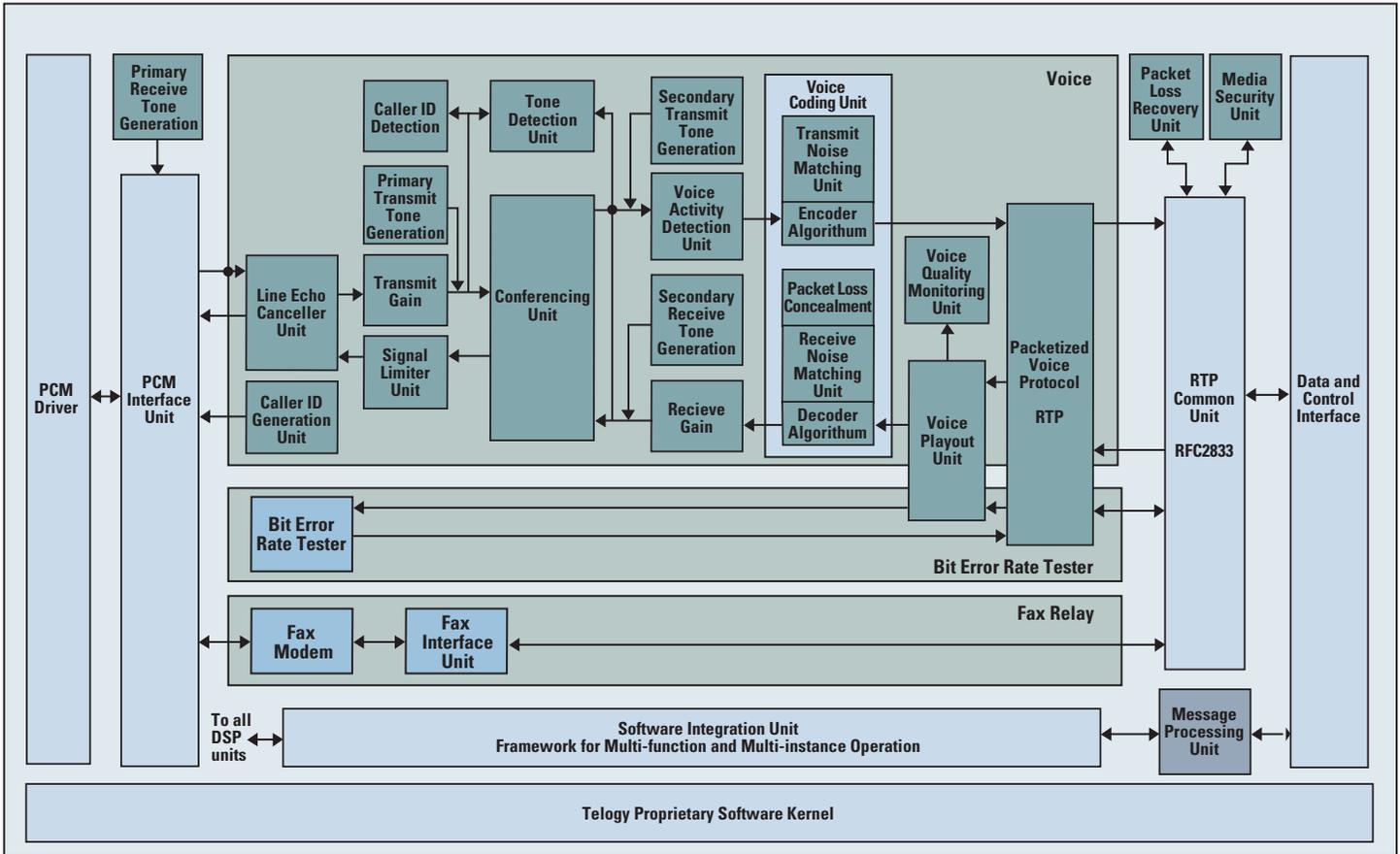


Figure 3

The echo itself is normally only about 6 ms to 8 ms in length and can occur at any time during the “echo tail.” The primary factor in determining how much echo tail is required for a specific application is the distance and delay expected for the voice signals – the longer the delay, the longer the echo tail. The delay is measured between the hybrid location and the speaker. The maximum echo tail length in use today is 128 ms.

Many business VoIP applications have similar echo canceller requirements as those found in infrastructure applications. With the TNETV2520, TI has integrated the Telinnovation echo canceller into the Telogy Software framework. This echo canceller module is G.168-2002 compliant and has been certified for use across carrier, enterprise and wireless telecommunications networks. Through a patented dual canceller algorithm, the Telinnovation echo canceller can converge and

suppress echoes within 25 ms at 90 percent of the time—much faster than echo cancellers that converge between 150 ms to 300 ms. For times when this is not possible, the Telinnovation echo canceller is comparable to other carrier-class echo cancellers. This same algorithm provides for superior double talk detection.

Voice Quality Monitoring, Manageability and Serviceability

Factors that impact voice quality in a VoIP network are fairly well understood. While most of these can be mitigated with careful network design, good quality assurance tools both in the VoIP endpoint equipment and the network itself can allow these issues to be addressed with the best balance of effectiveness and cost. The level of control over these factors will vary from network to network. This is highlighted by the differences between a well managed enterprise network vs.

an unmanaged network such as the Internet. In addition, scaling to very large networks increases exposure and places more importance on effective planning and implementation.

Voice quality is, in reality, the end users’ perception of quality. Telogy Software provides tools for VoIP network managers to better measure and manage quality in their networks. This extends the management capabilities of the industry leading Telogy Software solutions from TI to allow network managers to make in-service measurements of the voice quality their end-user’s experience, and to identify, localize, and address system issues affecting voice quality. The capabilities provided are based on TI’s unique experience providing VoIP systems, and these capabilities will continue to evolve as VoIP equipment and service providers continue to bring general VoIP capabilities to the market.

Modem over IP

With V.152 (Voice Band Data - VBD) voice band modulated signal samples are transported across an IP network using the RTP protocol. This requires a codec implementation that passes voice band modulated signals with minimal distortion. The VBD codec uses a specific RTP payload type that is negotiated with the remote V.152 implementation as defined in ITU-T Rec. V.152.

To support reliable VBD, the TI solution:

- Ensures end-to-end constant latency
- Disables Voice Activity Detection, Comfort Noise Generation and DC removal filters to improve modem data performance
- Exploits Forward Error Correction (FEC - RFC 2733) and Data Redundancy (RFC 2198) to improve modem data reliability (only if support has been successfully negotiated with the remote V.152 implementation)
- Exploits voice packet loss concealment techniques and algorithms suitable for modem and facsimile modulations

Packet Loss Recovery

High packet loss in managed IP networks rarely reaches the point where it affects voice quality. For applications over unmanaged networks (like the Internet) packet loss can easily have a serious affect

on voice quality. Techniques must be used to maintain voice quality in the presence of packet loss. Interoperability is achieved when these techniques are based on industry standards rather than proprietary techniques.

Implementations of RFC 2198 (Data Redundancy) and RFC 2733 (Forward Error Correction - FEC) help combat packet loss. In the Telogy Software, Data Redundancy allows for historical voice/data frames to be packed as redundant data as part of the current (primary) voice/data frame. When packet loss is detected, the data of the lost packet is retrieved from this redundant packet; when no packet loss is present, only the primary data is retrieved from the redundant packet.

For FEC traditional error correcting codes are used to compensate for lost packets. The FEC algorithms at the transmitter take the media packets as an input. They output both the media packets that they are passed and new packets called FEC packets. When packet loss is detected at the receiver the FEC packet are used to reconstruct the packet.

These packet loss recovery techniques have bandwidth and delay implications. Using Telogy Software, OEMs can build a complete voice quality management system by monitoring for packet loss, selecting the appropriate vocoder and packet loss recovery technique to optimize voice quality, bandwidth and delay performance.

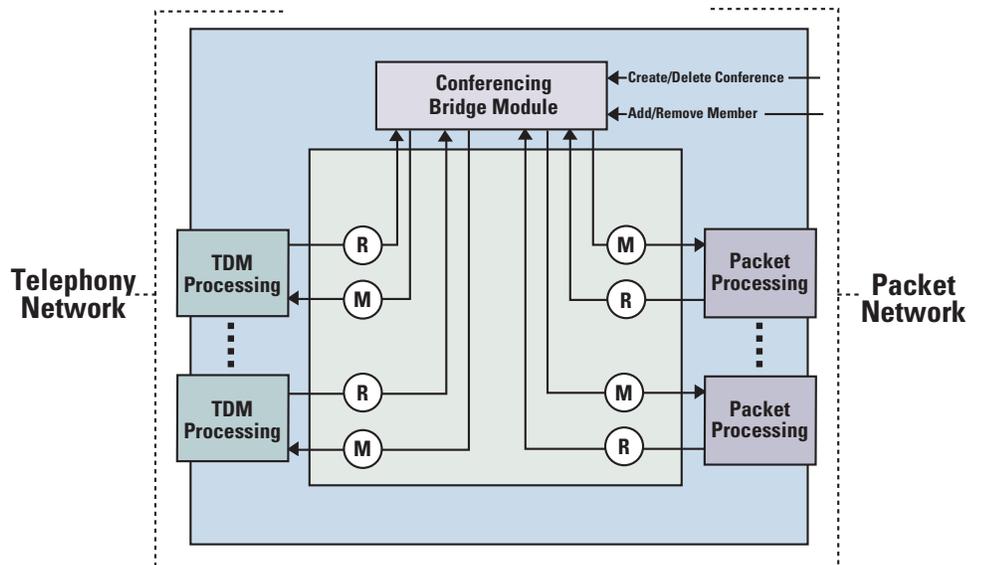


Figure 4

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