VFS-24/30/48/60
Single/Dual E1/T1 Digital Voice Compression Server Modules

- Compression of 24/30/48/60 digital voice channels
- User-selectable compression rates: 8 kbps (G.729A), 5.3 kbps (G.723.1), or 6.4 kbps (G.723.1)
- On-board framer supporting ISDN PRI or T1/E1 PBX trunks, with DSU, CSU/LTU interface
- Voice activity detection, silence suppression, and comfort noise generation for efficient bandwidth utilization
- Analog voice compression

VFS modules connect and compress E1/T1 PBX voice trunks for efficient transmission over IP or TDM networks. Each single-slot VFS module provides one or two E1 or T1 ports (external ports), enhanced by one or two additional internal ports for server functionality. The timeslots received from the E1/T1 trunks or from the Megaplex backplane are compressed using standard algorithms such as G.723.1 (6.4 or 5.3 kbps per channel), or G.729A (8.0 kbps per channel). Compression methods are user-selectable per bundle.

The modules also act as voice and fax relay compression servers, compressing analog voice channels within the same module. A basic application is shown in Figure 1.

VFS modules are available in the following versions:
- VFS-24 (single T1 trunk)
- VFS-30 (single E1 trunk)
- VFS-48 (two T1 trunks)
- VFS-60 (two E1 trunks)

Since the VFS modules can compress and transmit an entire E1/T1 trunk using as little as 3 timeslots, Megaplex can utilize the remaining main link bandwidth to provide additional services such as data, Ethernet LAN, POTS and/or management, all on one platform (see Figure 3).

Multiple VFS modules enable Megaplex to compress and transmit as many as 300/240 digital voice channels (10 full E1/T1 trunks) over a single E1/T1 link. This results in very efficient utilization of E1/T1 or SDH/SONET networks.

To facilitate point-to-multipoint applications, voice timeslots can be grouped together into up to 8 separate bundles (internal ports). Up to 30 (E1 module) or 24 (T1 module) channels can be connected to each internal port. Each internal port can be routed to a different main link or IP destination (if using ML-IP). Likewise, each internal port can operate at a different voice encoding rate and/or fax rate.
With regular TDM voice encoding methods alone, much bandwidth is "wasted" when normal periods of silence occur during a telephone call. VFS modules employ Voice Activity Detection (VAD) and silence suppression techniques, to maximize bandwidth utilization. After the G.723.1 or G.729A compression of the voice channels, silence suppression allows only channels transmitting actual conversation to fill main link timeslots; periods of silence are removed. At the remote side “comfort noise” is inserted to recreate the periods of silence that were removed, so that the quality of the call is not noticeably affected.

These methods use less bandwidth to transmit the same amount of voice without degrading the quality of the call. The user can select the amount of bandwidth (number of timeslots on the main link) that will be allocated per internal port, in accordance with the desired compression rate and the statistical periods of silence expected. In addition to voice transmission, the VFS modules also perform automatic fax relaying, which allows the transmission of Group III fax (ITU-T Rec. V.17 and V.29) at rates of 2.4 to 14.4 kbps, irrespective of the digitizing rate selected per voice.

The modules have automatic rate fallback capability, to automatically switch to the next lower data rate supported by both communicating faxes. Voiceband modem transmissions at all the standard rates up to 14.4 kbps (per ITU-T Rec. V.22bis and V.32bis) are handled in the same way as fax transmissions.

Modem transmissions are handled as voiceband data.

Voiceband modem traffic at all the standard rates up to 14.4 kbps (per ITU-T Rec. V.22bis and V.32bis) are handled in the same way as fax transmissions.

The VFS modules can act as voice compression servers for voice modules, as shown in Figure 2.

The audio processing performed by the digital signal processors (DSPs) includes:

- Voice processing
- Processing of DTMF and call progress tones
- Processing of fax and voiceband modem signals.

Each DSP processes its group of timeslots independently of the other DSPs, in accordance with the parameters selected by the user for that group. However, within each group, the DSP detects the signal type (voice, DTMF, fax, etc.) carried in each timeslot (channel), and automatically selects the appropriate processing method.

A built-in adaptive G.168 echo canceller cancels the near-end hybrid echo. The echo canceller improves the voice quality on voice lines with long delay, such as long-distance calls or calls over non-terrestrial links (e.g., satellite). Echo delays of up to 16 msec are tolerated.

Diagnostics include loopbacks towards local and remote PBXs, and a local loop per timeslot bundle.

### Specifications

**Module Versions**
- VFS-24: single T1 port
- VFS-30: single E1 port
- VFS-48: two T1 ports
- VFS-60: two E1 ports

**Number of Internal Ports (Timeslot Bundles)**
- Up to 8 ports for regular operation
- Up to 2 additional ports for server operation

**Number of Voice Channels (Timeslots) per Internal Port**
- T1 modules: up to 24
- E1 modules: up to 30

**Voice Encoding Rate**
-Selectable per internal port:
  - G.729A: 8 kbps per channel
  - G.723.1: 6.4 kbps or 5.3 kbps per channel

**Fax Rates**
- Group III rates, selectable per internal port: 2.4, 4.8, 7.2, 9.6, 12.0, 14.4 kbps

**Bandwidth Allocation on Trunk**
- Selectable, according to the programmed voice encoding rate

**Acceptable Channel Bit Error Rate**
- $1 \times 10^{-3}$ or better

**Adaptive Echo Canceller**
- Tolerates delays of up to 16 msec per channel, as per G.168

**Silence Suppression**
- G.723.1A, G.729B

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![Figure 1. Transport of Compressed Digital Trunks between Megaplex and Vmux](image-url)
### Signaling
CAS, including R2 and E&M
Transparent CCS

### Transparent Timeslot Mode
Selected timeslots can be transmitted transparently for data or management relay

### Timing Modes
**INT Mode:** Clock is provided by MP-2100 to the PBX
**LBT Mode:** Clock is provided by the PBX to MP-2100

### Diagnostics
Auto self-test activated upon power-up and during normal operation
- Local loopback towards the local PBX trunk
- Remote loopback towards the remote PBX trunk
- Local loopback on each internal port (bundle)

### Indicators
- **ALARM** (red) – On when fault detected on module
- **ON LINE** (green) – On when port is connected and functioning
- **TST** (yellow) – On when test being performed on port
- **LOC/Red Alarm** (red) – On when local sync loss is detected
- **REM (red)/Yellow Alarm** (yellow) – On when remote sync loss is detected

### Configuration
Via the Megaplex management system

### Power Consumption

<table>
<thead>
<tr>
<th>Module</th>
<th>Current [A]</th>
<th>Power [W]</th>
</tr>
</thead>
<tbody>
<tr>
<td>VFS-24</td>
<td>1.0</td>
<td>5.0</td>
</tr>
<tr>
<td>VFS-30</td>
<td>1.2</td>
<td>5.8</td>
</tr>
<tr>
<td>VFS-48</td>
<td>1.6</td>
<td>8.0</td>
</tr>
<tr>
<td>VFS-60</td>
<td>1.7</td>
<td>8.3</td>
</tr>
</tbody>
</table>

*Note: VFS modules utilize the +5 VDC line only.*

### Environment
- **Operating temperature:** 0°C to 45°C (32°F to 113°F)
- **Storage temperature:** -20°C to +70°C (-4°F to +160°F)
- **Humidity:** up to 95%, non-condensing

### T1 INTERFACE MODULES

#### Voice Channels
Up to 24 (per port)

#### Data Rate
1.544 Mbps (per port)

#### Standards
- AT&T TR-62411, Pub. 54016, fANSI T1.403, ITU-T Rec. G.703, G.704

#### Framing
D4 (SF), ESF

#### Line Code
AMI

#### Zero Suppression
Software-selectable:
- Transparent (AMI coding - no zero suppression)
- B7ZS
- B8ZS

#### Transmit Signal Level
Nominal level: ±2.7V (±10%), software-selectable, measured at 0-655 ft

#### Receive Signal Levels
0 to -10 dB without CSU

#### Timing
Internal or loopback

#### Jitter Performance
Per AT&T TR-62411

#### Line Type
Balanced 4-wire, 100Ω

#### Connectors (per port)
RJ-45

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![Figure 2. Transport of Compressed Digital and Analog Channels over TDM Network (VFS as Compression Server)](image)
**E1 INTERFACE MODULES**

**Voice Channels**
Up to 30 (per port)

**Data Rate**
2.048 Mbps (per port)

**Standards**
ITU-T Rec. G.703, G.704, G.732

**Framing**
G.732N
G.732N with CRC-4
G.732S
G.732S with CRC-4

**Line Code**
HDB3

**Transmit Signal Level**
Balanced: ±3V (±10%)
Unbalanced: ±2.37V (±10%)

**Receive Signal Levels**
0 to -10 dB without LTU

**Timing**
Internal or loopback

**Jitter Performance**
Per ITU-T G.823

**Line Type**
Balanced: 4-wire, 120Ω
Unbalanced: coax, 75Ω,
(jumper-selectable)

**Connectors (per port)**
Balanced: 8-pin RJ-45

**Note:** An MP-CBL-RJ45/2BNC/E1 adapter can be ordered to convert an E1 port RJ-45 connector into a pair of BNC connectors for unbalanced coax interface.

**Ordering**

**MP-2100M-VFS-24/T1**
Single T1 Digital Voice Compression Server Module for MP-2100/2104

**MP-2100M-VFS-30/E1**
Single E1 Digital Voice Compression Server Module for MP-2100/2104

**MP-2100M-VFS-48/T1**
Dual T1 Digital Voice Compression Server Module for MP-2100/2104

**MP-2100M-VFS-60/E1**
Dual E1 Digital Voice Compression Server Module for MP-2100/2104

**OPTIONAL ACCESSORIES**

**MP-CBL-RJ45/2BNC/E1**
Adapter for VFS-30/60 modules converting an E1 port RJ-45 connector into a pair of BNC connectors for unbalanced coax interface.

Figure 3. Combining Compressed Digital Voice, Analog Voice and Data in Point-to-Multipoint Topology