The JVC SUPER DIGIFINE series of hi-fi components are standard-setters that define the shape of things to come in audio. The series includes not only high-performance digital equipment but also components designed to take advantage of the dynamics of digital sound. What's more, as befit our top-of-the-line hi-fi components, each is finished in classy titanium gray and features exceptionally clean design.

Enjoy digital reality the JVC way—with SUPER DIGIFINE hi-fi components.
JVC's advancing D/A conversion technologies bring you more delicacy and subtlety

The heart of a digital processing system, like that of a CD player, is the D/A (Digital-to-Analog) converter. It's no exaggeration to say that it's the D/A converter that can make or break a CD player, determining the precision, accuracy and sound quality. That's why JVC offers two solutions to the problems that jeopardize the accuracy and precision of D/A conversion. Our K2 Interface eliminates "ripple" and "jitter" that can harm sound quality by transmitting digital codes, and codes alone, in their purest possible form.

The Quadruple Full-Time Linear 18-Bit Combination D/A Converter combines high resolution with the ability to precisely reproduce subtle nuances.
The K2 Interface: a revolutionary "pure" signal transfer system

Background
Once we assumed that so long as binary codes, or bits (1s and 0s), were correctly converted, stored, transmitted and retrieved, digital signals would not change, even in the slightest.

But studio engineers and artists involved in digital recording had a different story to tell—they found that there were changes in sound quality; changes that were noticeable when these recordings were mastered, pressed into a disc, and played back.

What our engineers have discovered is this: when a recording is dubbed from one tape to another, there is a change in sound quality, despite the fact that no analog processing is involved. Indeed, sound quality can change not only when tapes are dubbed but also when a different master tape is used, when a different master recorder is used, or when recordings are edited or equalized on a master console. This phenomenon occurs along the routes connecting various equipment whenever digital transmission is involved. So it's no wonder that the master tape and the discs produced from it sound different, or that the discs made from a master recorded in one country sound different from the discs produced in another.

Audio engineers are also finding that using different digital connection cables between the CD player and an amplifier can cause differences in sound quality. And the same thing happens when a digital code is transmitted inside the CD player from the digital section to the analog. It also occurs inside amplifiers that come with built-in D/A (Digital-to-Analog) converters as the digital signal is sent from the digital to the analog section; that is, at the interface between the digital filter and D/A converter.

The quality of digital sound, therefore, does change during processing—but why? We discovered the answer through a collaboration of engineers in JVC's musical industries and a professional digital recording system developing staff in which they explored the intricate digital world of binary codes. The result of this research is the K2 Interface—a revolutionary signal transfer system for pure sound reproduction.

The implications of the K2 Interface are great, because it can be applied in digital audio and other digital equipment, whether the application is professional or consumer oriented. As a matter of fact, it can be used as a universal interface for digital equipment.

Ripple and jitter
As a digital signal is transmitted from the digital to the analog section, non-code components—components that are totally unrelated with the transmitted code—are generated inside the digital section and passed on to the analog section. It is these components, we have discovered, that cause the change in sound quality.

These unwanted non-code components are called ripple and jitter. Ripple is waveform distortion, whereas jitter is a shift in time. They are generated by load fluctuations in the power supply or stray capacitance and inductance of the circuits. As you can see in the figure above, non-code components are superimposed over bits for 1s and Os and passed to the analog section for processing. Specially, a signal with ripple and jitter contains, in addition to the basic clock frequency, high-frequency harmonics that should not be there.

Moreover, digital circuits generate noise when their devices are in a transient state—switching in and switching out. This type of noise is generated because the change in voltage and current occurs in a finite time. All this means that a digital signal contains both harmonic distortion and noise. When this signal is transmitted to the analog signal section, it causes various types of interference and adds intermodulation distortion in the analog signal. A digital signal may contain only a small amount of non-code components, but the distortion they cause contains harmonic components that are totally unrelated to the original analog signal. Therefore, in excessive cases, this type of distortion can even change the nature of digital music.

Concept of conventional digital signal transmission
Non-code components (ripple and jitter) are passed to the digital signal processing section during transmission, affecting the transmitted signal and hence greatly changing the sound quality.
Conventional optical transmission system

A conventional optical transmission system uses a waveshaping technique to remove ripple and jitter. This system is supposed to cut electric interference between the digital and analog sections, eliminate ripple and jitter, and provide "clean" signal transmission, but in-use tests have shown that it doesn't. True, the transmitted signal looks clean on the scope, but it actually contains ripple and jitter—spurious harmonics that change the waveform. Once inside the analog section, these harmonics interfere with the grounding system and mix into the analog output, ultimately affecting sound quality.

The K2 Interface: the concept

At JVC, we tackled the problem of ripple and jitter by devising a new angle. The idea was to devise a design whereby codes, rather than waveforms, would be transmitted, and a totally new code would be generated by the receiver. In this way, non-code components can be eliminated from the signal transmitted to the analog section.

In the K2 Interface, the digital and analog signal processing sections are completely separated so that spurious non-code components cannot be transmitted from one section to the other. This system has a "transmitter" and a "receiver." In this design, the receiver looks at the transmitter at regular intervals for a brief enough period to know whether the code being sent is a 1 or a 0, and then generates the appropriate code on its own. That's the idea behind the K2 Interface.

With the K2 Interface, coded digital signals are optically transmitted through a photocoupler from the digital processing section to the analog processing section. A code detection switch is connected to the ground of the photodetector, and is normally open. Therefore, the digital and analog sections are completely separated, making it impossible for non-code components to enter the analog section. Digital codes are controlled by the sync signal generated by the master clock in the analog section. Sampling moments are dictated by the timing of the master clock. The code is detected at each half-code length, and the detected signal is not affected by ripple, jitter, or noise. Since the code is detected at each half-code length, the detected signal is not affected by jitter. And since the switch is closed for only the briefest of moments—just 20 nanoseconds—the signal is not affected by ripple, either. Therefore, the spurious non-code components are not transmitted with the signal. Finally, the detected code is sent to a D-type flip-flop and a new code—708.6 nanoseconds in length—generated according to the timing of the master clock. The output from the flip-flop then goes to the digital filter, the D/A converter and eventually to the analog circuitry.

The K2 Interface optically decouples the digital from the analog section, improving performance by K2 Interface

In the data above, it appears that the transmitted code by the K2 Interface is free of ripple and jitter.

Application

The K2 Interface optically decouples the digital from the analog section, improving performance by K2 Interface. Here is proof of how the K2 Interface removes ripple and jitter, reduces intermodulation distortion and noise.

The graph frequency is contained in a modulator, and in the output of the modulator, it is modulated into a signal in the frequency band of the modulator. As you can see, the noise is mixed into the output signal. The output signal is a signal that is modulated onto a carrier frequency in the frequency band of the modulator. The output signal is a modulated signal.

Improved performance by K2 Interface

In the data above, it’s apparent that the transmitted code by the K2 Interface is free of ripple and jitter.
The graphs on page 4 show the frequency spectra of residual noise contained in the analog output from a model with and without the K2 interface. In this test, heavy loads are applied to the power supply for the transmitter in each model in order to induce non-code components—random noise—into the FM transmission signal.

As you can see, when random noise is mixed with the digital signal in a model without the K2 interface, it increases the residual noise level and changes its overall shape. This means that external disturbances applied to the analog section can affect the analog output and cause distortion so severe that it can alter the nature of the music. In the model featuring the K2 Interface, residual noise remains the same even if random noise is mixed with the digital signal. This is proof of the K2 Interface's ability to shut out external disturbances and prevent them from affecting the analog output.

Applications of the K2 Interface

We have used the K2 Interface in our Compact Disc Player—the XL-2101TN. But because of its universal nature, it can also be applied in other digital equipment as well, like amps with built-in D/A converter (our AX-Z100TN is one), DTR (Digital Tape Recorder) decks, direct satellite broadcast reception device professional equipment—indeed anywhere digital transmission is involved.

The digital sound processed by the K2 Interface is distinguished by better resolution, enhanced ambience and higher sense of realism it presents. We assure you that the K2 Interface you will be enjoying the studio-quality digital sound at home very soon.

Advantages of 8-times oversampling digital filtering

The digital code read by the pickup from a Compact Disc is sampled by frequency of 44.1 kHz and processed for 16-bit requantization. This process generates not only audible frequencies but also spurious frequencies whose spectrum occupies the area just above 20kHz, the higher end of the audible range. These are frequencies that are multiples of the sampling frequency with a difference of ±20kHz or ±40kHz—24kHz or 64kHz, for instance. These spurious frequencies are usually eliminated by a low-pass filter with a sharp attenuation response and complex design to prevent the noise they can create. But too-sharp attenuation inevitably compromises sound quality.

In the XL-Z1010TN, we use a digital filter that re-samples the digital signal at a frequency 8 times the original—that is, 352.8kHz instead of 44.1kHz. This shifts unwanted frequencies to a much higher frequency range than normal. The advantage of using such a high sampling frequency is that the low-pass filter used in combination can have a gentler attenuation response. This means phase distortion is drastically reduced to provide you clear imaging of stereo sounds. Moreover, it eliminates requantization noise more effectively.

By using a higher re-sampling frequency, our digital filter reduces the sampling rate from about 22 microseconds to only a few. This improves the resolution in the time domain as well as conversion accuracy. Our digital filter internally processes 16-bit data with double precision accurate down to the 32nd bit. Then, before the processed data is output to the D/A converter, it's rounded to 18 bits for the subsequent D/A converter.

JVC's Quadruple Full-Time Linear 18-Bit Combination D/A Converter

The D/A converter ultimately reconstructs the oversampled digital code from the digital filter and converts it into an analog signal. For precise D/A conversion we developed the "Quadruple Full-Time Linear 18-Bit Combination D/A Converter" to operate in tandem with an 8-times oversampling filter.

Our advanced D/A converter features two D/A converter units for each channel—four in all. There is a 16-bit converter for most significant bits and a 2-bit converter for the two least significant bits. Since the least significant bits have greatest bearing on the sound quality at low levels, we use an elaborate discrete D/A converter system for these bits to ensure higher precision. Currents from these two converters are summed channel by channel, and the converters operate with 18 bits "full-time" whether the level is high or low.

By combining an 8-times oversampling digital filter and 18-bit combination D/A converter, the digital signal processing circuit inside the XL-Z1010TN improves level resolution and time-domain resolution by 4 times and 2 times, respectively, over digital processing circuits using a 16-bit D/A converter with a 4-times oversampling digital filter.

All in all, our Quadruple Full-Time Linear 18-Bit Combination D/A Converter improves linearity, precision and resolution, allowing you to enjoy powerful and highly pure digital sound both at their most delicate or dynamic.
Designs for quality digital sound

The digital signal processing section isn’t the only part in the XL-Z1010TN we redesigned to improve sonic reproduction. We also took pains to improve other sections—including the analog section—in order to further refine overall performance. And the unit is mechanically constructed to prevent resonance, vibration and interference.

New high-precision laser pickup design

The XL-Z1010TN features a newly designed 3-beam laser pickup that combines high sensitivity, precision, stability and immunity to resonance and vibration. It’s precise because the distance between laser beams has been shortened. And it’s stable and resistant to resonance and vibration thanks to JVC’s new suspended actuator. Our pickup is also compact and lightweight, improving tracking accuracy and reducing the noise caused by servo-controlling currents.

Digital outputs—optical and coaxial

The XL-Z1010TN is equipped with two digital outputs—both optical and coaxial—so you can directly interface it with external digital components (e.g., the D/A converter in your amplifier or digital signal processing equipment). Using the optical output will electrically insulate the two connected components, shutting out digital noise that can compromise signal quality. What you get is purer, more musical digital sound.

Disc stabilizing clamper

Now that 3-inch (8cm) singles are available along with 5-inch (12cm) discs, the servo control in a CD player is given the new task of compensating for differences in weight, or moment of inertia, between the two disc types by adjusting servo currents. This process can generate noise. Therefore, we’ve added a large stabilizing clamper to the disc rotating mechanism to equalize the moment of inertia for both types of discs. The clamper improves speed stability and keeps variations in servo currents and noise to a minimum.

Separate digital/servo and analog circuits

In most CD players, digital and servo-control circuits are laid out together with analog circuits. So the noise that servos generate during active operation or the click pulse noise that microcomputers and digital circuitry create can easily affect the performance of analog circuits. In the XL-Z1010TN, however, we’ve separated the digital and servo-control circuits from the analog circuits, both physically and electrically. As a result, interference between digital and analog circuits is no longer a problem, longer aff. Moreover, fluorescent lights will not create a problem, either.

New Y Servo for superior tracking ability

Our “New Y Servo System” uses two special tracking beams—one leading and one trailing the main beam. The time difference between these signals is compensated for and the two signals are compared so as to cancel each other out. It is this sophisticated processing that enables the pickup to remain locked on the correct track—even when the disc is dirty or scratched. So the XL-Z1010TN’s pickup does not “skip” tracks or repeat the same track over and over.

Double-floating Independent Suspension System

Subtle vibrations and resonance can degrade digital sound by affecting parts and devices inside a CD player. Therefore, in the XL-Z1010TN, the pickup base is suspended from the mechanism base, the mechanism base from the chassis—a design we call the “Double-floating Independent Suspension System.” The result is that the pickup tracks a disc with highest accuracy despite shocks and sound pressure from speaker systems, thus ensuring clean reproduction unsullied by resonance and vibration.

Disc stabilizing clamper

Now that 3-inch (8cm) singles are available along with 5-inch (12cm) discs, the servo control in a CD player is given the new task of compensating for differences in weight, or moment of inertia, between the two disc types by adjusting servo currents. This process can generate noise. Therefore, we’ve added a large stabilizing clamper to the disc rotating mechanism to equalize the moment of inertia for both types of discs. The clamper improves speed stability and keeps variations in servo currents and noise to a minimum.

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Designs for ease of use

Another hallmark of the JVC XL-Z1010TN is its user-friendliness, for, after all, half the pleasure of dynamic digital audio is its convenience. And nowhere is this more evident than in our state-of-the-art CD player.

3-way edit function: auto, programmed and multi-disc

The 3-way edit function is a boon to those who are active in editing and recording CD tracks on tape. The "auto edit" mode lets you automatically fill a tape with as many selections from a disc as the tape can hold. Specify the length of time that one side of a tape can accommodate ("45" for a C90 tape, for instance), and the tracks are automatically recorded on tape–first side A and then B–in the same order as on the disc. The "programmed (manual) edit" mode lets you program tracks you want to transfer from disc to tape in any order. And as you do, you can assign the side on which each track will be recorded, A or B. And the "multi-disc edit" mode permits you to even program tracks from a number of discs for transfer to tape, assigning the side of the tape on which each track will be recorded. Using any of these modes, you can keep the unused blank space on a recorded tape to a minimum.

Motor-driven volume control

The XL-Z1010TN comes with a full-function remote control. As volume is adjusted from the remote, a motor is activated to drive the volume potentiometer in the main unit to adjust playback level. Its smooth, low-noise operation will not compromise sound quality as electronic volume controls often do. And there are two outputs on the rear panel—one with fixed and one with remote-variable level—for your operating convenience.

Other features

- JVC COMPU LINK Control System for interactive operation
- Multi-function display with 20-track program chart
- Headphone output and line output with volume control
- Random access programming of up to 32 tracks
- In-scan
- 4-way repeat (one track, all tracks, programmed tracks, A-B)
- Random play, skip and search
- Auto/manual search
- Remote control with volume control and numeric keypad
- Controls behind swing-down panel
Intelligent design and a better user interface improve hi-fi operation

The RX-1010VTN Receiver from the JVC SUPER DIGIFINE Series provides you with a look at the shape of a receiver in which computer control has been pushed to its logical limits. Most of its controls and operations are handled by microcomputers accessed via the handy remote control. Therefore, electronic switches are relegated to behind the swing-down door. And in the RX-1010VTN, our COMPU LINK Control System is represented in its most advanced and most convenient form, highlighting the new CSRP (COMPU LINK Source-Related Presetting).
COMPU LINK Source-Related Presetting (CSRP):
Acoustic memory program by program

Up until now, you have had to manually adjust levels, balance, equalization, and other parameters for the best sound each time you switch from one program source to another—touching up lows and highs as you change inputs from CD to FM, for example. But thanks to JVC’s new COMPU LINK Source-Related Presetting (CSRP), you can program settings for the following parameters program source by program source or even for each of 40 FM/AM stations.

Volume: You can preset the volume program by program or station by station. This means the slight differences in sensitivities between components can be compensated for to ensure uniform output.

Balance: If you’ve finely adjusted channel balance for the best in your room, it can be committed to memory program by program.

Equalization: You can recall a preset or user-programmed equalization and store it in memory by program source or station. In this way, you can use a flat (192kHz) response for digital sound and a response with boosted ends for FM, for instance.

Surround: The surround mode and adjusted parameters may be preset for each program source or each preset station.

Loudness: The on/off position of the loudness switch can also be programmed source by source.

DAP (Digital Acoustics Processor): You can assign one of seven available ambiance types (SYMPHONY HALL, RECITAL HALL, OPERA HOUSE, CHURCH, LIVE CLUB, STADIUM and MOVIE THEATER) to each program or each preset station.

Suppose that in setting these parameters, you’ve seen to it that when a video program is selected, volume is set at a moderate level; balance is set at center; an equalization for MOVIE is selected; Dolby Pro-Logic is turned on; and the "MOVIE THEATER" is set for digital acoustics processing. Then, press the button, you can be immediately transported into a world of incredible wrap-around sound. And this, of course, holds true for other inputs like CD, PHONO, TAPE and even each of 40 FM/AM preset stations.

In addition, to improve ease of use, chosen parameters and digits are temporarily displayed on the fluorescent panel to help you check with the "CSRP TEST" or "CSRP DISPLAY" function and, if necessary, change the set parameters.

COMPU LINK Communications System (CCS): For better rapport between man and machine

The CCS is another enhancement added to our COMPU LINK Control System, providing a better visual way to communicate between the user and the receiver. It graphically displays the mode of operation of a connected component in play, even for CD players and turntables. And it lets the user give a custom name, up to five characters long, to each FM or AM station or equalization that is preset and put into memory—"JAZZ-1" and "AM-3" for stations, "DISCO" and "POPS" for equalizations, for instance. Also shown are the chosen program source, volume level in digits, name of the Digital Acoustics Processor effect recalled from memory, and a host of other useful information. All you have to do to check the status of the receiver is just look at the display panel.

Example of Fluorescent Display with CSRP Preset (VCR-1 as the source)

(1) Source selected

(2) Volume, balance, loudness setting

(3) D.A. graphic EQ setting

(4) Dolby surround setting

(5) DAP setting

(6) Normal setting
Programmable Remote Control—the ultimate in hands-off operation

The clean looks of the RX-1010VTN owe much to the fact that many of the functions are electronic and therefore are consigned to the handy remote control. Moreover, this remote is a programmable one which can learn functions of other infrared remote controls.

Up until now, it has been necessary to keep handy all of the remote controls for individual audio and video components—a situation which tends to invite confusion. But our programmable remote control will replace nearly all the remotes for your audio and video components. It comes preprogrammed to control the receiver it comes with, of course, but also selected JVC components, like decks, CD players (even CD auto-changers), turntables, VCRs and TVs.

And our programmable remote control also has the ability to learn control codes of most other infrared remote controls made by other manufacturers. It stores the control codes for a total of 180 functions (should each use JVC's standard code length) in memory, allowing you to operate your audio and video components from a single unit.

In addition, our programmable remote comes with an LCD (Liquid Crystal Display) panel. As you select the desired program source, the display on the panel changes to show an appropriate menu screen showing only functions you need. Two modes are offered: the "standard" mode with 11 menus and the "programmable" mode with 14. For even more convenience, the symbols and numbers in each menu are not for display alone; they are actually touch-activated control buttons.

When program sources are changed, the display on the panel also changes to show related functions.

**Light button**
Provides back-lighting to improve readability in low-light situations.

**Touch-panel LCD display**
This panel shows all the functions you've programmed for a particular input, but hides the functions you don't need. Displayed "keys" are actually touch-activated control buttons.

**Indicators**
Indicators in this section simplify operation by telling you when a signal is sent or a function is learned, or when an error occurs during operation. Also featured here are a number of buttons for functions such as display screen pattern selection (SELECT 11), CSRP on/off (CSRP/CANCEL), component power on/off, and remote on/off.

**Display buttons**
Change the display without changing the selected program source.

**Source selection buttons**
When program sources are changed, the display on the panel also changes to show related functions.

Programmable Remote Control replaces most others, giving you the power to control nearly every remote-control component—regardless of make—in your integrated audio-video system. To program it, just flick one of its switches to "LEARN," align our remote and an existing remote end-to-end, and press corresponding buttons on each. That's all!
**RX-1010VTN**

**RECEIVER**

**SYSTEM CONTROL CENTER**

You the remote-less of video lack one of our remote to-end, and s on each.

**Digital Acoustics Processor**

The Digital Acoustics Processor, a technology based on our advanced digital engineering and our exclusive digital signal processing expertise, is built into the RX-1010VTN. It's a digital means of creating the ambience of a musical space—auditorium, concert hall, etc.—at home by applying a delay to sound. This system allows you to get the same exciting 3-dimensional sound you would at a live performance. Seven patterns of sound fields are resident in memory (SYMPHONY HALL, OPERA HOUSE, CHURCH, LIVE CLUB, STADIUM and MOVIE THEATER), each available at a touch. You can also fine-adjust the ambience by controlling acoustic parameters such as ROOM SIZE, LIVENESS and WALL TYPE, so the created ambience sounds most realistic in your own listening room. JVC's precision computer-controlled sound field analysis system called the "systematical 6-point sound field analysis method" and channel-by-channel acoustics processing guarantee you'll get a truly thrilling "you-are-there" feeling.

**Dolby Pro-Logic Surround Sound**

The RX-1010VTN is equipped with the latest Dolby Surround sound decoder—the Pro-Logic Surround sound decoder. Using a new adaptive-matrix sound steering circuit, it enhances the sense of direction by boosting the output from the dominant channel and reducing the output in non-dominant channels. It also clearly localizes the dialog at the center channel, so conversation naturally comes from the screen. With the new Pro-Logic Dolby Surround, channel separation has been dramatically increased from mere 3dB to 25dB. Moreover, the RX-1010VTN's decoder uses digital delay circuit, therefore the circuitry will not degrade sound quality. All in all, the new depth and width you hear from video tapes and discs will add a new dimension to your video watching.

**Dynamic Super-A for class-A sound**

Dynamic Super-A is a refinement of our original Super-A technology. With Super-A, a certain amount of bias, or idling current, is constantly applied to the power transistors to prevent them from switching off. The smooth, sinusoidal waveforms it provides are proof of the reduced harmonic distortion. Because Super-A does not generate switching distortion, it lets you enjoy low-distortion class-A sound. Dynamic Super-A further reduces distortion while improving the amp's overall response. Our Gm Driver has also been added to improve performance under actual in-use conditions.

**Output power:**

- **Stereo:** 120 watts per channel, min. RMS, both channels driven into 8 ohms, from 20Hz to 20kHz with no more than 0.007% total harmonic distortion.
- **Surround:** 110 watts per channel, min. RMS, both channels driven into 8 ohms, from 20Hz to 20kHz with no more than 0.007% total harmonic distortion (front); 15 watts per channel, min. RMS, both channels driven into 8 ohms, at 1kHz with no more than 0.07% total harmonic distortion (rear).

**Block Diagram of Dynamic Super-A with Gm Driver**

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**LOGIC**

Digital Acoustics Processor

Sound Field Analysis Patterns

Digital Acoustics Processor

**RX-1010VTN**

RECEIVER/SYSTEM CONTROL CENTER

superDIGIFINE

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By just adding an extra pair of speaker systems you can instantly upgrade to exciting live ambience sound.
High dynamic power

Today, more dynamic power is demanded of amplifiers than ever, because of the wider dynamic range and better transient response of digital programs. So in the RX-1010VTN, we've improved the power supply so that it can instantly generate as much current as needed, especially when the impedance drops to 4 ohms or lower. Another way we've improved dynamic power is through the use of high-power, high-performance output transistors. The high dynamic power of 360W x 2 at 2 ohms that our receiver can provide is an indication of how it's ready to drive low impedances as presented by the combination of quality speakers and high-quality program sources.

Superb ease of use

Over the years, the functions of our receivers have become more and more complex while their versatility has improved immensely. With the RX-1010VTN, we've simplified operation with the help of computers, electronic control and remote control—the reason why the front panel of the receiver is clean and uncluttered, despite its awesome control capabilities.

Computer-controlled 7-band S.E.A. graphic equalizer

The JVC S.E.A. graphic equalizer is a popular choice among audiophiles. With it you can change the sound to suit your own tastes, to overcome the acoustic deficiencies of your room, or to create a special sound for headphone or in-car listening. This is because a graphic equalizer allows you to adjust limited frequency ranges without affecting others. Besides, JVC S.E.A. graphic equalizers are known for accuracy, low noise and distortion and wide dynamic range.

We've built a 7-band graphic equalizer into the RX-1010VTN to give you the immense power of sound equalization. And to make it easier to use, we've computerized its operation. Here's what computerized equalization does for you.

Example of 7-band graphic equalizer indications (Frequency and control range)

<table>
<thead>
<tr>
<th>Setting</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>160 Hz</td>
<td>14 dB</td>
</tr>
<tr>
<td>1 kHz</td>
<td>6 dB</td>
</tr>
<tr>
<td>5 kHz</td>
<td>6 dB</td>
</tr>
</tbody>
</table>

Computer-controlled tuner

By combining a microcomputer with a PLL-quartz digital synthesizer tuner, the RX-1010VTN is superbly easy to use and accurate.

- Preset memory: You can preset a great total of 40 FM/AM stations for instant recall at the touch of a button.
- Auto memory: You can have the receiver preset 40 FM/AM stations all automatically for you.
- Preset scan: All 40 preset FM/AM stations in the memory can be automatically sampled for about 5 seconds each, helping you find the station you want.
- Station name display: You can assign a name of up to five alphanumeric characters to each preset station—"JAZZ" and "ROCK L", for instance—for easy identification.
- And remember: The CSRP lets you customize acoustic parameters (equalizer settings, balance, volume, etc.) and keep them in memory, station by station.

Computer-controlled 7-band S.E.A. graphic equalizer

The JVC S.E.A. graphic equalizer is a popular choice among audiophiles. With it you can create and store five custom equalizations, giving each the five-letter name of your choice (even your own name!).

5 "nameable" user-preset (manual) equalization: You create and store five custom equalizations, giving each the five-letter name of your choice (even your own name!).

5 programmable equalizations: Five equalization patterns called HEAVY, CLEAR, SOFT, MOVIE and VOCAL are resident in memory.

Reverse equalization: You can reverse an equalized response at a touch, say, to "compand" (compress/expand) a tape recording to reduce hiss noise.

Remote control: With a remote in hand, you can equalize the sound and recall a preset equalization from where you sit—for the ultimate in equalizing ease.

Information

A highly visible display lets you see RX-1010VTN extremely in indicator functions. logical/linear control remote video three unique

Audio/Video

The RX-1010VTN is an extremely versatile audio/video receiver with comprehensive features. It can handle a variety of video sources including composite, component, and S-video signals. It also includes a built-in 7-band graphic equalizer and a computer-controlled tuner for easy, accurate operation. The receiver is designed to be user-friendly with a clean, uncluttered front panel. It is a great choice for anyone looking for a high-quality, high-performance audio/video receiver.
RX-1010VTN
RECEIVER/SYSTEM CONTROL CENTER

Informative large-sized display
A highly visible, large fluorescent display keeps you informed of the function you've chosen with the RX-1010VTN. The display is extremely easy to read, because all indications are consolidated into logical groups.

Audio/video integration
The RX-1010VTN is ready to integrate your audio and video components into one easy-to-use home entertainment system—conveniently controlled from a single remote.

Video connections: Up to seven video components (VIDEO, VCR-1 and VCR-2) may be connected and switched in and out. You can also dub from disc to tape or tape to tape. Moreover, there are S-video terminals in parallel that have separate pin contacts for chroma (C) and luminance (Y). By connecting TVs and VCRs (S-VHS VCRs) with matching terminals, you can enjoy a better picture and more accurate color.

Monitor output: A monitor output lets you connect a monitor TV. There's an additional S-video terminal for a monitor with matching separate-chroma/luminance terminals.

Sound selector: The selector lets you mate the audio of your choice with any video, a handy feature when you make your own video productions or “ambience” videos.

Other features
- High-gain phono equalizer for MM/MC cartridges
- Connections for 2 pairs of speaker systems
- Sleep timer
The digital way to bring the live musical experience home

Recorded performances you hear on your stereo system sound fine, especially if they're digitally made, but the same piece of music played by the same artist sounds more exciting and more engrossing when you hear it live. But this truly realistic sense of "being there" is hard to reproduce on a conventional hi-fi system. The JVC XP-A1010TN Digital Acoustics Processor, however, is by no means conventional: it lets you enjoy the incredibly realistic ambience of a live concert right in your own home.
What’s “liveness”? Suppose you’re in a hall listening to a live musical event. When a note is played, the first thing you hear is the direct sound from the source. Direct sounds are then followed by “early reflections”—a group of sounds that are reflected by the walls, ceiling and stage with delayed volume. Finally what reaches your ears last are the “reverberations” which come from random directions over a relatively extended period. It’s this pattern of sounds, reflections and reverberations that lets you know you’re experiencing a live performance.

Each acoustic space (a hall or auditorium, for instance) has its own individual ambience, or the pattern of reflections and reverberations. When a space is large and acoustically “live,” for example, intervals between individual early reflections are rather long, and reverberations decay slowly. The opposite is true for a small-sized room with “dead” acoustics.

Digital acoustics simulation When a signal is turned into digital form, it can be delayed or otherwise processed without its quality being compromised or degraded. And computers have the amazing ability to perform complex analyses of a vast amount of data at very high speeds thanks to efficient VLSIs (Very Large Scale Integrated Circuits).

With this combination of digital technologies, we can now generate reflections of a concert hall with exceptional accuracy—by measuring precisely the acoustic components in the sound heard in a hall, determining the direction and level of each, and then recreating this complex aural environment in the listening room, all electronically and digitally. In this way, it is possible to create an incredibly realistic ambience right in your own listening room.

But while digital technology has given us the awesome ability to turn a listening room into a concert hall, there are two important factors we must be aware of. One, the measured data on the acoustic characteristics of actual halls that will go into the memory of a digital acoustics processor must be accurate and comprehensive; otherwise, precise reproduction is impossible. And two, to synthesize the ambience of a hall accurately, two things must be considered—the ambience contained in the recorded music and the ambience of the listening room where the recorded music will be played back.

JVC symmetrical 6-point sound field analysis method

To obtain accurate data on the ambiances of existing halls, we first developed a computerized system to measure them and quantify them—the “symmetrical 6-point sound field analysis method.” With this system, a starter’s pistol is used as the source to create the direct sound, since the noise it generates is pulse. It’s shot in a hall, and the sound and the reverberations it creates are picked up by a JVC-developed device using three pairs of precision-calibrated omnidirectional microphones set along X, Y and Z axes, equidistant from the reference point. The signal picked up by the microphones are digitally recorded by JVC’s Digital Audio Mastering System, and the components that comprise the picked-up sound are instantly shown on a display, with the resulting data stored on a floppy disk for later analysis.

Typical pattern of sound

Impulse response

Acoustic response of a musical space
In actual measurements, the sound source is placed at the center stage, and the microphones are set at a seat where a listener would get the best sound. The same measurements are repeated with the sound source placed to stage left and right, and the pickup system is placed at the best seat in the house. Unlike conventional measuring systems, ours can measure the ambience of a small space, like that of a listening room. This has made it possible for us to compensate for the ambience of the listening room in the Digital Acoustics Processor.

To collect data that would go into our Digital Acoustics Processor, our engineers took the trouble of traveling all over Europe to measure the acoustics of famous halls there using our elaborate measuring system. We went to this trouble because we know that the simulation of a hall cannot be any more accurate than the data on which the simulation is based. The data on the sound source and reverberations measured by the symmetrical 6-point sound field analysis method are displayed as 3-dimensional patterns for closer examination—each of the reverberations represented as a "virtual image source" along with its size and location. In this data, the center shape represents the floor of a hall. The largest circle is the direct sound, while others are "virtual image sources"; the farther away from the center, the longer delay they are given. Level is expressed by the size of a circle.

Features of the Digital Acoustics Processor

In order to accurately synthesize the ambience of a hall at home, our Digital Acoustics Processor compensates for spurious ambiences that are generated as recorded music is played back in the listening room—the ambience contained in the recorded music and the ambience of the listening room itself. This is because the "virtual image sources" created by our digital processor will add their own reflections by bouncing off the walls, floor and ceiling in the listening room. So these must be first cancelled out. The same is true of the reverberations contained in the recorded music if they are not fully compensated for; however, excessive reflections and reverberations can be added, totally ruining the sense of realism. Also our digital processor gives the user the choice of "solo" or "spread" according to the size of the musical source, to recreate the ambience of a hall more precisely. The "LISTENING ROOM SIZE," "LISTENING ROOM REVERB," "SOURCE REVERB," and "SOURCE POINT/SPREAD" controls allow the user to adjust these parameters to get a desired ambience in any listening room from any program source. Further, the JVC Digital Acoustics Processor contains 2 different measured ambiances memory. This lets you instantly recreate ambiances of some typical musical spaces. And, to let you alter a pattern you've chosen more to your liking, it has controls to adjust six parameters for a desired ambience—ROOM SIZE, LIVENESS, LOW PASS FILTER, REVERB LEVEL, HIGH FREQUENCY REVERB, and OFFSET DELAY.

In our Digital Acoustics Processor PCB, the flowchart shows the function of digital processing, including listening room source in synthesizing ambience.
The process of simulating half ambience

In our Digital Acoustics Processor, a digital-processing VLSI (Very Large Scale Integrated Circuit) has the function of synthesizing the "virtual image sources" by digital processing for playback in a user's listening room. Virtual image sources in a "target" hall are compared with the data for the target hall—"if it is duplicated, it will be removed.

The process will be repeated for the 2nd largest reflection component in the hall, the 3rd largest, and so on, within the limits of the hardware. Into this loop of eliminating spurious ambience components is incorporated the environment set by parameters for each of the listening room, the sound source and the target hall (LISTENING ROOM SIZE, LIVENESS, etc.).

Moreover, in our Digital Acoustics Processor, digital processing of ambience is performed in stereo rather than in mono as in the past. By making better use of the ambience components contained in a source—including out-of-phase left/right information—it's possible to create the ambience of the target hall more accurately and provide a fuller musical experience.

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Purity, dynamics and an enhanced sense of power

What are the two most critical criteria demanded of an amplifier in this age of digital audio? High power and high purity. That's because it's the combination of these two factors that lets you enjoy digital sound to its fullest. Until now, getting both has been next to impossible. But now JVC brings you both high purity and high power with the help of sophisticated digital engineering and our Digital Pure-A II circuitry, both featured in the AX-Z1010TN integrated amplifier from the SUPER DIGIFINE series.
Digital Pure-A II

High purity vs. high power

The look of reproducing music with dynamic power and delicate purity is a daunting one for an amplifier. Of course, today's amplifiers cannot give you the full impact of live music, but we're getting closer. There's one way to ensure high purity in a hi-fi system—by using a class-A amplifier. The reason is that this design does not suffer from 'switching distortion.' The one drawback of class-A amplifiers, however, is that they are inefficient, requiring a massive power supply and wasting most of it in the form of heat. So up until now class-A operation and high power could not exist in the same amplifier.

Push-pull configuration

Most amplifiers use an output configuration called 'push-pull' in order to improve linearity and efficiency of power transistors and also to achieve high power. In the push-pull design shown above, the transistor on top and the one below individually amplify the positive and negative side of an input signal to improve efficiency. The signals amplified by the top and bottom power transistors are combined in the output, and, driven by the power supply, are turned into power to move the speaker.

The power supply plays a critical role in any amplifier, especially if it's designed for better digital reproduction. The supply should feed enough power to the power transistors so that the transistors operate with high efficiency and reproduce both dynamic and delicate sounds with power to spare. In this respect, remember power (W) equals current (I) times voltage (E). This means, if the current is given, you'll have higher power by increasing voltage, and vice versa. From this it's apparent that a reduction in power will result if there's loss in either current or voltage.

Class-A vs. class-B operation

An amplifier can be either class-A or class-B, depending on how the transistors in the push-pull arrangement amplify the applied signal. With class-B amplifiers, each of the paired transistors amplifies an input while it is conductive, but is switched off as it tries to amplify the side it's not responsible for. Because little bias is applied, this results in switching and crossover distortion, which impairs purity. Besides, the waveform of varying current is not as smooth as a sine wave. This means the output contains the kind of harmonic distortion that can be irritating to our ears. Yet the advantage of class-B amps is that since little bias is applied, excessive heat is not generated, so the amps do not need a bulky and costly heat-sinking system. For the same reason, since they are highly efficient, they can easily deliver a high power. In class-A amps, a sufficiently high bias is constantly applied to the power transistors in order to accommodate higher dynamic levels. This ensures that the current waveform is much smoother and that the combined output looks closer to that of the input. This input/output symmetry is maintained at any level so that pure reproduction is achieved with extremely low amounts of harmful harmonic distortion. But since a high level of bias is applied to the transistors, excessive power is generated and much is dissipated as waste heat. This means a high-power class-A amp requires a prohibitively costly heat sinking system and power supply.

So all in all, what's the most ideal amplifier design? It's the concept of the two classes of amplifiers without disadvantages of each—one that combines the low distortion and smooth response of class-A with high power and efficiency of class-B.

JVC Digital Pure-A II

The JVC Digital Pure-A II operates just like the most ideal amplifiers, offering the combined advantages of class-A and class-B. It's an amplifier that varies the bias according to dynamics of the input in order to combine high purity and high power: it supplies low bias when the input level is low, intermediate bias when the level is moderate, and higher bias when the level is high. In other words, Digital Pure-A II is an ideal amplifier that combines high power, high efficiency and pure sound. The innovative thing about this design is that it takes advantage of the unique nature of the digital signal in digital form, a signal can be delayed without degrading its quality. Digital Pure-A II operates in the following way:

Digital signals fed directly from digital equipment (a CD player, for example) are split into two signals: the main and the 'prediction' signals. The main signal is sent to a time base processor where it's stored in memory for a very brief 10 milliseconds before being passed on to the D/A (Digital-to-Analog) converter. The prediction signal is sent to a prediction signal processor and on to the high-speed optical bias control in the prediction signal processor. The level of the upcoming main signal is measured, and a prediction signal is generated. In the optical bias control a control signal is generated by comparing and
analyzing the prediction signal and the D/A-converted main signal from the "output level detector," where the absolute peak level of the power transistors is detected. Finally, the control signal is sent to the bias circuit to determine which of three bias steps is to be applied to the power transistors.

If there is no input, only minimum amounts of currents are applied (step 1). (In the figure above, the lower trace is the level of the upcoming signal, and the upper trace shows the level of bias current.) When the level, or dynamics, of the input signal increases to a moderate value—and this is the level you hear music at most of the time—the current increases to step 2, allowing you to enjoy the benefits of pure class-A operation. When the input signal level goes higher still, the current increases to the third, highest level at which you can enjoy both high power and pure class-A sound.

In Digital Pure-A II, rather than the power-supply voltage, the bias current fed to the power transistors is adjusted according to the level of the input signal, which translates into quicker response and higher efficiency. The bias current is always high enough to ensure that the amp operates in pure class-A, which reduces harmful harmonic distortion and provides pure sound. And the power-supply voltage for the power transistors is high enough to avoid clipping the waveform when a high-level signal is suddenly applied.

Output power: 100 watts per channel, min. RMS, both channels driven into 8 ohms, from 20Hz to 20kHz with no more than 0.004% total harmonic distortion.

The AX-Z1010TN is a "digital" amplifier complete with a digital signal processing circuit for accepting and processing digital signals from your CD player, DAT (Digital Audio Tape) deck or other digital equipment. Here we’ve used the same circuit configuration as in our state-of-the-art XL-Z1010TN CD player: the K2 Interface, 8-times oversampling digital filter, and the Quadruple Full-Time Linear 18-Bit Combination D/A (Digital-to-Analog) Converter. In essence, the K2 Interface removes ripple (waveform distortion) and jitter (shifts in time) by creating a totally new code. The oversampling digital filter removes re-quantization noise more effectively, while reducing group delay. And our exclusive combination D/A converter improves low-level linearity in order to reproduce delicacy and subtlety better. More details of these JVC Technologies are found in the description of the XL-Z1010TN CD Player.
Opt Super-A

The AX-Z1010TN lets you enjoy class A sound from analog electronic switches that let us avoid running connection wires from the back of the chassis to the front and then back again. Moreover, there is a "DAC DIRECT" switch that when turned on allows the output from the D/A converter to go directly to the power amp after a very short trip, with only the volume control on line.

Low-impedance driving capability

The dynamic sound of digital programs requires that power supply current be always available to cope with low impedances presented by high-power transistors and complex loads. This ensures that digital programs are reproduced with low distortion and power to spare. The AX-Z1010TN is a "digital" amplifier for a reason: it's designed to handle load impedances as low as 4 ohms or even less. Indeed at 2 ohms it provides dynamic power as much as 320 watts per channel. We've made this possible in two ways. One, we've used high-performance, high-power transistors for the output devices and arranged them in parallel push-pull configuration. This reduces output impedance for improved speaker-driving capability and reduced distortion. Two, we've used thicker wiring in the power transformer to reduce power-supply impedance and improve the dynamic range.

Large power transformer using thick wire and high-capacity capacitors for the power supply

AX-Z1010TN is a "digital" amplifier for a reason: it's designed to handle load impedances as low as 4 ohms or even less. Indeed at 2 ohms it provides dynamic power as much as 320 watts per channel. We've made this possible in two ways. One, we've used high-performance, high-power transistors for the output devices and arranged them in parallel push-pull configuration. This reduces output impedance for improved speaker-driving capability and reduced distortion. Two, we've used thicker wiring in the power transformer to reduce power-supply impedance and improve the dynamic range.

Other features

- High-gain phono equalizer for MM/MC cartridges
- Bass control
- Remote control with volume control and speaker switching
- Connections for 2 pairs of speakers

Lightweight, high-power transistors in parallel push-pull configuration.

This reduces output impedance for improved speaker-driving capability and reduced distortion.
FX-1010TN
COMPUTER-CONTROLLED
DIGITAL SYNTHESIZER
TUNER

Computer control at its most accurate and convenient

The primary function of a tuner is to bring in the station you want precisely and accurately, whether it's near or far. Using the latest circuit designs and devices, we've made our tuner from the SUPER DIGIFINE series, the FX-1010TN, more sensitive, selective and interference resistant than ever—the reason the tuner is able to provide wide dynamic range, low noise and low distortion, and wide frequency response. It's also extremely easy to use, thanks to the amazing power of an advanced microprocessor.

Reception servo for optimum reception

The reception servo in the FX-1010TN ensures the best reception from any station, almost anywhere. A built-in microprocessor detects the strength of a tuned station and compares it against the degree of interference from adjacent stations. Then, depending on the degree of interference, the microprocessor selects the optimum operation mode for front-end, IF and multiplex decoder stages (adjusting such parameters as RF gain, IF bandwidth, Quieting Slope Control and mono/stereo). Therefore, when interference is excessive, a wide IF bandwidth is automatically chosen to prevent noise. And if there's no interference, then a narrow IF bandwidth is automatically selected to give you remarkably clear sound. When a signal level is overly strong, the RF gain is reduced to avoid saturation distortion; when it's weak, sensitivity is increased.

Moreover, the tuner is equipped with inputs to connect two antennas; each may be oriented for the best reception of stations in diametrical opposition to the other. Consequently, the position of antennas, A or B, may be stored in memory station by station, which allows most precise reception from any station, without multipath distortion. To add to convenience, the selected parameters are clearly indicated on a large fluorescent display, letting you quickly confirm tuning status.

Features for low noise and low distortion

JVC uses components and devices to ensure lowest possible noise and distortion and widest dynamic range from your favorite stations.

"Opticalink" system

The "Opticalink" system is one developed by JVC to ensure highest purity from your favorite stations. It uses eleven photocouplers (each consisting of a photodiode and photoemitter) to electrically decouple the analog from the digital section. As a result, interference between the two due to electric coupling is completely eliminated. This puts an end to digital noise and removes any trace of muddiness from the sound you hear.

In the FX-1010TN, each of the analog/tuner, digital control and "Opticalink" sections is mounted on its own PC board to shut out mutual interference and noise, and the digital control section, which can be a source of noise generation, is fully shielded to contain noise.
Inside view of the FX-1010TN

Computer-controlled operating ease
With the help of powerful microcomputers, we've also improved tuning ease of the FX-1010TN tremendously.

Station name display
You can assign up to six alphanumeric characters to each preset station in the memory—"JAZZ-8," for instance.

Auto memory
All 40 FM/AM stations can be automatically tuned in sequence and committed to memory as presets.

Random preset memory
You can preset as many as 40 FM/AM stations in random order. There's a numeric keypad that allows direct access to any of 40 stations.

Preset scan
All preset FM/AM stations can be automatically sampled one by one for approximately 5 seconds each.

Preset cancel
Use this feature to skip undesired preset stations during preset scan.

Program memory
Up to eight "events" (broadcasts) can be programmed for sequential recall under the control of an optional timer.

Auto QSC
The Auto Quieting Slope Control circuit automatically goes into action to reduce noise when a station signal is weak.

dB-referenced signal strength indicator
Read off the signal strength of a tuned station accurately down to 1 dB—a convenience when orienting antennas.

Variable stop level
Adjust FM/AM muting threshold in 5 dB steps over a range during auto tuning. The variable stop level feature lets you adjust the threshold from 20 dB to 80 dB for FM, and from 60 dB to 90 dB for AM. When you use a higher level, you'll receive only powerful, clear-sounding stations, with weak stations muted out. Or, when you choose a low level, all receivable stations are tuned in.

Record calibration signal generator
Record calibration signal generator outputs a standard 400 Hz signal for recording level adjustment. So, you can easily set the recording level for different broadcast types of tape.
Getting closer to digital sound in performance

It's true that audio equipment has been becoming more and more elaborate in circuitry these years. But the more complex circuitry is, the greater there are chances that the signals will be compromised in the process. So in the TD-V1010TN cassette deck from the SUPER DIGIFINE series, JVC took the classic "less is more" approach. Also in order to ensure pure and clean sound, JVC has employed a number of ways to damp and eliminate resonance and vibration.
"Direct" design for pure reproduction

Based on the basic design philosophy of "less is more," the TD-V1010TN features minimum wiring and a signal path as direct and straightforward as technically possible.

"DIRECT" inputs

The TD-V1010TN has a "CD DIRECT" input—and a switch to select it—that lets you route the analog signal from a CD player directly to the cassette deck, entirely skipping circuitry in the receiver or amplifier. The TD-V1010TN also has a "DIRECT" input for connection of a second high-quality program source and the normal "LINE" input for connection with the amplifier, both with associated selector switches.

With "CD DIRECT" and "DIRECT" inputs, the applied sign, r runs the shortest route skipping the balance control. Even the Doby HX-Pro and BIC noise reduction circuits can be completely put off line with the NR switches off. Trimmed wiring and fewer contacts add up to exceptional purity in the played back sound.

Direct layout

Also, the PD board for recording and playback is directly connected with input and output terminals, and the pattern is laid out symmetrically for the left and right channels to trim wiring. In addition, the input selector buttons, input level knob and power button on the front panel are linked via remote bars with switches and controls located near the input and output terminals on the rear. This of course is to cut the wiring to the bare minimum.

All this care to the last detail has resulted in reduction of noise, distortion and other sources of sound degradation that can compromise the purity of taped sound. This in turn means you can enjoy the fuller benefits of digital sound from our cassette deck—wide dynamic range and accurate recreation of most minute nuances, to name a few.

Higher resistance to vibration and resonance

Even the slightest vibrations and resonance can affect the quality of taped sound: when the tape is subjected to vibration during recording or playback, it can lead to modulation noise, the source of unclear, impure sound. So we've devised a number of ways to damp, suppress or totally eliminate them, ensuring pure reproduction.

Vibration-suppressing mechanical designs

In the TD-V1010TN, the precision base for the tape drive mechanism is built from rigid and heavy die-cast aluminum. The front panel is fashioned out of non-resonating high-specific-gravity resin, which is 15 times heavier and 1.7 times more rigid than the conventional front-panel material. During recording and playback, the cassette is held firmly in place by a stabilizing pad that damps resonance and vibration. A metal sheet is attached to the circuit board supporting the motor to stop the flywheel from vibrating. Further, a massive brass weight is used for a reel receptacle to damp resonance and also improve rotating constancy. A solid and rigid base supports the entire chassis. Large insulators isolate the deck from sound pressure and external vibrations. And control knobs and switches are fashioned out of solid aluminum. This all adds up to pure and clean sound, especially from digital sound.

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Closed-loop dual-capstan drive with direct-drive motor

The TD-V1010TN employs perhaps the most sophisticated tape drive mechanism ever designed. With this drive, the portion of tape that runs across the heads is held taut during both recording and playback. One advantage of this sophisticated system is that it shuts out vibration and other external disturbances, resulting in reduced modulation noise. This gives you sound that is noticeably cleaner and clearer. And it improves head-to-tape contact for better response. The capstan is driven by a JVC precision high-torque coreless "pulse servo" motor which does not set up much vibration, further "cleaning" the taped sound.

"Acoustic modulation noise" up until now vibration and the degree of sonic degradation it can cause have not been correlated as meaningful data. So JVC has devised a new and more useful way to measure the effects of vibration on performance—the "acoustic modulation noise" test. In conventional modulation noise tests, the deck under test is set to simply record and play a 10kHz signal. But with our new method, the test deck is set to do so under 100-phon sound pressure, generated by a speaker reproducing pink noise. This setup simulates more closely the situation in which a deck is likely to be used at home.

As you can see below, "acoustic modulation noise" is not only reduced overall in the TD-V1010TN, its curve is smooth. This proves that the TD-V1010TN has the capability to deliver exceptionally clean and transparent sound.

3-head design for best response

The TD-V1010TN features three independent heads, one each for recording, playback and erase. This means since the heads are separate, each can be given the optimum gap width for best response. Moreover, the azimuth, tilt and tracking can be aligned head by individual head for best results—extended response and low noise. This discrete 3-head construction also enables you to monitor recordings as they are being made. There are separate circuits for Dolby encoding and decoding, so you can monitor Dolby-encoded recordings in real time.
"Fine" amorphous heads using OFCCC wiring

The record and play heads in the TD-V1010TN are made of newly developed "fine" amorphous material which combines superior high-frequency response and our SA (Sid-Akoy) head's superior resistance to wear. For higher signal purity and clarity, we use PCCOC for the coils and leads in the heads. PCCOC is purest oxygen-free copper with single-crystal-like construction, featuring more efficient signal transmission and coloration-free sound reproduction.

Dolby* HX-Pro circuit

Dolby HX-Pro is a new approach to improved high-frequency response. During recording, this circuit dynamically adjusts the bias according to the dynamic level of music in order to improve the tape's saturation level and expand the dynamic range at high frequencies. In combination with Dolby B/C noise reduction systems, it allows a tape to record a wide range as wide as that of digital programs. With most other decks, Dolby HX-Pro is undefeatable. But you can pull it off line with the TD-V1010TN whenever you want higher purity.

High bias frequency

The bias frequency is twice higher than normal at 210kHz. This reduces beat noise due to interference of audio signals with the oscillator frequency.

Low-impedance voltage-tracking power supply

A highly regulated, low-impedance power supply is featured in the TD-V1010TN. Using a large-capacity transformer and electrolytic capacitors, it automatically tracks positive and negative voltages to keep the ground potential zero and improve the stability of amplifier circuits. Moreover, our power supply exhibits extremely low output impedance across the audible frequency range. The result? Even most dynamic musical passages or external noise cannot affect the steady performance of the power supply.

Separate circuit construction

In the TD-V1010TN, the audio amps, power supply and control circuits are mounted on separate PC boards, and insulated from each other, to shut out interference—both electric and magnetic. Furthermore, the deck uses circuit boards plated with OFC (Oxygen-Free Copper) to ensure that the most delicate nuances are reproduced precisely. Also contributing to cleaner sound is a fluorescent display off switch which turns off the display to prevent low-level digital noise.

Other features

• Computer-controlled full-logic control with "silent mechanism"
• Fluorescent Digital Peak Display, level meters and digital counter
• Auto rec mute, Music Scan, Timer start (record/play), Auto tape selector
• Bias and level controls for flat response and sensitivity matching
• Headphone output with volume control

"Dolby" and the double D symbol are trademarks of Dolby Laboratories Licensing Corporation.
Another step closer to the "digital realism"

For highly musical reproduction, especially of digital programs, high power alone is not enough. Therefore, in our speaker system for the SUPER DIGIFINE series, the SX-911WD, we've also improved linearity (especially at low levels), clarity, transparency, definition and depth. We have done so by upgrading our speaker-design technologies, from diaphragm unit to construction, to give you exciting "digital realism."
Cloth-carbon woofer

Our new cloth carbon woofer provides extended bass response, openness and richness—properties thought to be mutually exclusive until now. This is because the newly developed cloth carbon, carbon fiber woven like cloth, features the ideal combination of high weight, high rigidity, high speed of sound and controlled internal loss.

Cloth carbon's high rigidity means that the vibrations of the voice coil are more exactly translated into the linear movement of the diaphragm, while the power range over which the diaphragm excites a piston-like motion (without cone breakup) is wider. Its controlled internal loss ensures an improved frequency response throughout the entire range the woofer covers. And the light weight means that the upper limit of the unit's frequency range is higher, extending the overall low-frequency response. Moreover, the surround of the woofer is made of a pint material featuring quick release, which helps make the bass sound powerful, crisp and rich. The midrange also features "Fire" cloth carbon for clear and natural sound.

Amorphous-diamond coated tweeter

The amorphous-diamond coated tweeter provides exceptional transparency. This is because we use a totally new design for the tweeter: a dome diaphragm with a titanium base on which a thin layer of amorphous diamond is coated using a high-tech process called CVD (Chemical Vapor Deposition). Featuring uniform thickness, high purity and smoothness of surface, this coating increases the diaphragm's speed of sound to almost that of natural diamond. So the transient response is dramatically improved, as is sonic purity.

Die-cast aluminium speaker frames

Every unit in the SX-911 WD is housed inside a solid, unresonating die-cast aluminum frame. All frames are circular to disperse vibrations uniformly and efficiently. The woofer frame is especially impressive: it weighs 4.6 lbs. (2.1 kg)—twice the weight of a common woofer frame. Firmly mounted by eight heavy-duty screws on the front baffle, the frame supports the moving structure so that rich and powerful bass of digital programs is played back with highest clarity. The solid frame also contributes to increasing the rigidity of the front baffle, protecting it from resonance and vibration. This ends spurious radiation, which means dramatic clarity in the reproduced sound.
A solid enclosure with natural sonority

The SX-911WD features a solidly constructed enclosure, using 1-inch (25mm) panel boards. To increase rigidity, front and rear baffles are mounted with additional cleats. And all sides of the enclosure are bonded together under one-ton pressure, making the entire enclosure as strong as it were made of a single piece of wood. This bonding process does not use heat, so the boards will stay firmly bonded for years, maintaining the enclosure’s high rigidity. Moreover, the boards are made from high-density particleboard, chosen for its superb musical sonority.

Rounded front baffle

The round-cornered front baffle of the SX-911WD does more than lend class to overall design. It prevents the sound diffraction that can occur at sharp edges causing blurry and indistinct sound imagery. You'll enjoy clear definition, smooth response, and accurate phase response from the SX-911WD.

Computer simulations

With JVC, the days of using trial-and-error methods to find the optimum positions for mounting speaker units on baffles are gone. Today, by inputting design parameters such as speaker unit characteristics, the physical properties of enclosure materials, and the sizes and shapes of front baffles, we can realistically simulate on computers how the sound will be generated and propagated. Thanks to this advanced technique, the SX-911WD combines better definition, smooth response and accurate phase response.

Quality parts

We use only quality parts in the SX-911WD so that the sound, whether digital or analog, is the best you have ever heard. For instance, the frequency-dividing network is made from quality parts to minimize signal loss and phase distortion. Moreover, the network is also physically divided into three parts—one each for the high, mid and low frequency range—to shut out interference. Furthermore, wires are not soldered but clamped firm and then hermetically sealed by a bonding agent. This ensures minimum degradation of the signal over longer periods of time.

- High power handling capacity: 150 watts/300 watts (Music)
- 12-inch (30.5cm) cloth carbon woofer for the bass sound that's crisp, extended and rich
- 4-1/2-inch (11.5cm) "fine" cloth carbon midrange for rich and natural midrange sound
- 1-inch (2.5cm) amorphous-diamond coated tweeter for transparency and superior transient response
- Frequency response: 40—50,000Hz
- Sensitivity (1m on axis): 91dB/W

Propagation Characterisics of SX-911WD

Vibration Analysis by Modal Technique

Thanks to thick boards and rigid construction, the enclosure of the SX-911WD is highly resistant to even the minutest deformation and resonance.
SX-911WD
THREE-WAY SPEAKER SYSTEM
SUPER DIGIFINE

paclty jusc carbon and that's it cloth and

}-- 916B/W
SPECIFICATIONS

**XL-Z1010TN**
Compact Disc Player

**RX-1010VTN**
Receiver

**AMPLIFIER SECTION**

- Output Power: 200 watts per channel, min. RMS, both channels driven into 8 ohms from 20Hz to 20kHz, with no more than 0.05% total harmonic distortion.
- 4-Channel Operation: (Front/Center/Rear/Sub) 4-channel, both channels driven into 8 ohms from 20Hz to 20kHz, with no more than 0.05% total harmonic distortion.
- Dynamic Range: 97dB
- Signal-to-Noise Ratio: 90dB
- Total Harmonic Distortion: 0.007%
- Crosstalk: 80dB
- Power: 15 watts per channel, min. RMS, into 8 ohms, 1kHz, with no more than 0.01% total harmonic distortion.
- Frequency Response: 20Hz - 20kHz, ±0.5dB
- Signal-to-Noise Ratio: 90dB
- Total Harmonic Distortion: 0.007%
- Crosstalk: 80dB
- Power: 15 watts per channel, min. RMS, into 8 ohms, 1kHz, with no more than 0.01% total harmonic distortion.

**VIDEO INPUTS/ OUTPUTS**

- Input: 200mV (47k ohms) into 8 ohms.
- Output: 380mV/47k ohms, 20Hz - 20kHz, with no more than 0.007% total harmonic distortion.
- Frequency Response: 20Hz - 20kHz, ±0.5dB
- Crosstalk: 80dB
- Power: 15 watts per channel, min. RMS, into 8 ohms, 1kHz, with no more than 0.01% total harmonic distortion.

**FM TUNER SECTION**

- Selectivity: 3000/100Hz/1kHz
- Selectivity: 3/100Hz/1kHz
- Signal-to-Noise Ratio: 50dB
- Stereo: 15-3/10kHz (50Hz - 20kHz)
- Dimensions: 17-6/16" x 43/8" x 17-35/64" (440mm x 101mm x 435mm)
- Weight: 27.6 lbs. (12.5kg)

**VP-A1010TN**
Digital Audio Processors

- Output Power: 200 watts per channel, min. RMS, both channels driven into 8 ohms from 20Hz to 20kHz, with no more than 0.05% total harmonic distortion.
- Dynamic Range: 97dB
- Signal-to-Noise Ratio: 90dB
- Total Harmonic Distortion: 0.007%
- Crosstalk: 80dB
- Power: 15 watts per channel, min. RMS, into 8 ohms, 1kHz, with no more than 0.01% total harmonic distortion.
- Frequency Response: 20Hz - 20kHz, ±0.5dB
- Signal-to-Noise Ratio: 90dB
- Total Harmonic Distortion: 0.007%
- Crosstalk: 80dB
- Power: 15 watts per channel, min. RMS, into 8 ohms, 1kHz, with no more than 0.01% total harmonic distortion.

**VIDEO INPUTS/ OUTPUTS**

- Input: 200mV (47k ohms) into 8 ohms.
- Output: 380mV/47k ohms, 20Hz - 20kHz, with no more than 0.007% total harmonic distortion.
- Frequency Response: 20Hz - 20kHz, ±0.5dB
- Crosstalk: 80dB
- Power: 15 watts per channel, min. RMS, into 8 ohms, 1kHz, with no more than 0.01% total harmonic distortion.

**FM TUNER SECTION**

- Selectivity: 3000/100Hz/1kHz
- Selectivity: 3/100Hz/1kHz
- Signal-to-Noise Ratio: 50dB
- Stereo: 15-3/10kHz (50Hz - 20kHz)
- Dimensions: 17-6/16" x 43/8" x 17-35/64" (440mm x 101mm x 435mm)
- Weight: 27.6 lbs. (12.5kg)
AX-Z1010TN
Amplifier

- Overall Characteristics

Output Power
100 watts per channel, non-RMS, both channels driven into 4 ohms from 20Hz to 20kHz, with no more than 0.04% total harmonic distortion.
100 watts per channel, non-RMS, both channels driven into 8 ohms at 1kHz, with no more than 0.07% total harmonic distortion.

Dynamic Headroom
- 23 dB for 1 kHz, 20 Hz to 20 kHz
- 20 dB for 1 kHz, 20 Hz to 20 kHz

Frequency Response
- 20 Hz to 20 kHz (±1 dB)

Signal-to-Noise Ratio
- 89 dB (20 Hz to 20 kHz)

Hum
- 20 Hz to 22 kHz, ±3 dB

Power Supply
- 115 V-220 V, 50 Hz/60 Hz

Dimensions
- 33.5 x 435 x 120 mm
- 13.2 x 17.1 x 4.7 inches

Weight
- 3.11 lbs (1.41 kg)

FX-1010TN
Tuner

- Antenna Input Impedance
  75 ohms unbalanced

- Output Signal Level
  800 mV (20 kHz)

- REC Gain Output Level
  Equivalent to 50% FM modulation

- Amateur Section

- Usable Sensitivity
  25 mV/m (10 dB S/N)

- Total Harmonic Distortion
  0.3%

- Dynamic Range
  60 dB

- Image Response Ratio
  45 dB

- IF Response
  20 dB (VHF, UHF, LF, AM)

- Dimensions (W x H x D)
  17.5 x 4.9 x 13.4 inches

- Weight
  8.2 lbs (3.7 kg)

TD-V1010TN
Cassette Deck

- Head: Fixed Configuration, Fine Anodized Heads for Recording and Playback

- Motors: Pulled belt, deduced motor

- Input Sensitivity/Impedance: 800 mV/100k ohms (Direct input)

- Output Level: 3.5 V RMS (600 ohms)

- Dimensions (W x H x D)
  22.7 x 17.1 x 4.3 inches

- Weight
  22.7 lbs (10.3 kg)

- Features
  - Head/track level, whiteface, without MT. The S/N is improved by about 2dB at 550Hz and about 20dB above 1kHz with Dolby-C on.
  - Strobe
  - 4-way, automatic reverse

SX-911WD
Speaker System

- Impedance
  8 ohms

- Frequency Range
  40 - 18,000 Hz

- Dimensions (W x H x D)
  6.5 x 3.4 x 5.9 inches

- Weight
  6.2 lbs (2.8 kg)